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PhD. Thesis

Contributions to the Future Media Internet using Service-Oriented Architectures

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“We can't solve problems by using the same kind of thinking we used when we created them.” (Albert Einstein)

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Abstract

Nowadays, video streaming applications are the most bandwidth-hungry applications and this tendency is envisaged to grow exponentially. With the proliferation of multimedia capable devices, multimedia services have to deal with heterogeneous environments where very different types of terminals wish to receive content anywhere and anytime. This situation motivates the appearance of multimedia services that adapt contents to the specific context of users. These services can benefit from the use of different technologies for content delivery (e.g. Peer-to-Peer and Network Coding), media signalling (e.g. SIP and P2P protocols), media representation (e.g. MPEG-7 and MPEG-21) or multimedia scalable and robust codification (e.g. Multiple Description Coding and Scalable Video Coding). However, current Internet architecture is based on a rigid layered model (TCP/IP-based) following the, no longer valid, end-to-end argument, which makes difficult to introduce new functionalities efficiently. To solve this, Service Oriented Architectures (SOA) principles seem to fit in the proposal of new architectures for a more flexible Future Internet based on services that can be invoked when and where necessary.

The objectives of this PhD. Thesis are exploring and validating different mechanisms for enabling Future Media Internet communications. To achieve this, we apply the SOA paradigm to provide efficient context-aware multimedia communications in the Future Internet.

This work proposes solutions to enable the seamless provisioning of multimedia services in the Future Internet by means of context-aware service discovery and composition processes which are integrated in a novel service-oriented clean-slate architecture. One goal is to provide adapted and personalized services, dealing with high dynamic and heterogeneous environments. For this reason, this thesis includes research on novel media coding techniques (Multiple Description Coding, Scalable Video Coding), Forward Error Correction codes, and distribution techniques (Peer-to-Peer, Network Coding) that can be applied to achieve seamless media communications. Moreover, context-aware service composition will address the requirements of media services (and any service in general), access methods, devices and interactions.

This work presents a radical view of the Future Internet, where the necessary functionalities for accomplishing communications, in user devices, in the network and at all levels are considered as services. Services are not fixed but dynamically composed where and when necessary, with respect to user service requirements, network transfer capabilities and surrounding context in the user and the network environments.

Composition of basic network-level services calls for a clean-slate approach to the Internet, while composition of higher level (transport and application) services prompts for an evolutionary approach. Nevertheless, composition of communication services manifests itself as a revolutionary way of looking communications and building communication systems.

This PhD. Thesis introduces two main architectural innovations clearly beyond current state of the art. Firstly, a Service-Oriented framework able to deal with (existing) functionality at all levels (connectivity, transport, application) by considering the provided service and not the technology behind the functionality. All these service functionalities can be seen as services thanks to suitable service-oriented abstractions that allow including existing functionality/protocols as well as new functionality in a flexible way. Secondly, we present a novel service-oriented clean-slate architecture generalizing Information-Centric Networking (ICN) approaches. This thesis would propose the first clean-slate architecture completely aligned with the work done within the ISO Future Network working group.

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Part I: Introduction – The Basics

Chapter 1 Motivation: Towards Future Media Internet

There is no doubt these days that the Internet epitomises a cornerstone tool for human communications but, at the same time, it has triggered new challenging problems. Amongst other demands, users expect higher levels of performance, security and reliability. Internet is growing beyond its original expectations and its fundamental design goals based on the end-to-end argument. As a result of its gigantic growth, the Internet is reaching some technological and operational limits imposed by its architecture in its attempt to give full support to the new requirements introduced by the increasing number of new services, applications and contents. However, whilst it still seems unclear how the current Internet architecture will handle these new requirements, the fast-growing need for new strategies in the network that can guarantee a certain level of Quality of Service (QoS) and Quality of Experience (QoE) appears to be at this stage paramount.

One of the most significant reasons for the fast growth of the Internet comes from multimedia and, concretely, from video. Current Internet traffic is increasing very fast mainly due to the proliferation of media capable devices, media services and their demand by Internet users. Digital production techniques are rapidly reshaping the industries for media production and broadcast. They have a tremendous impact also on the networking industry because of the huge number of networked media deployment and end user multimedia capable devices. Some analyses [1][4][20] confirm that video traffic will keep growing at a tremendous pace. Next, we give some numbers and predictions in Internet video growth.

- Internet video is now 40% of consumer Internet traffic, and will reach 62% by the end of 2015, not including the amount of video exchanged through P2P file sharing.

- The sum of all forms of video (TV, video on demand, Internet, and P2P) will continue to be approximately 90% of global consumer traffic by 2015.

- Internet video to TV tripled in 2010. Internet video to TV will be over 16% of consumer Internet video traffic in 2015, up from 7% in 2010.

- Video-on-demand (VoD) traffic will triple by 2015. The amount of VoD traffic in 2015 will be equivalent to 3 billion DVDs per month.

- High-Definition (HD) video-on-demand will surpass standard definition by the end of 2011. By 2015, high-definition Internet video will comprise 77% of VoD.

Thus, Internet requires efficient methods for accelerating the sharing and reliable distribution of this big volume of digital media according to different users need and context. Emerging digital TV services offer contents in multiple ways and different formats. Popularity of video services, such as Youtube or Zattoo, have made video traffic in the Internet the most present one. Here, Peer-to-Peer (P2P) systems play a crucial role as they allow distributing multimedia contents in a scalable and robust manner, overcoming the limitations of other technologies such as multicast by creating overlay networks. Even, today, there are appearing new revolutionary transmission techniques, such as Network Coding, which aim at introducing novel paradigms to increase the throughput and reduce the overall delay of communications whilst proposing an alternative to classic routing and avoiding only store and forward functions of intermediate nodes. However, these kinds of techniques are difficult to introduce in the current Internet in an efficient manner. They can be deployed at application level, but, in so doing, they cannot offer the most of their advantages.

Moreover, the heterogeneity of media capable devices connected to the network is also rising (e.g. handheld, PC, TV, tablet PC, smartphones, etc.) and it usually requires the creation or the adaptation of services and resources specifically for each target platform. This situation leads to too generic and static systems that do not provide the content adapted to the final device or too complex systems that require a big effort in development and maintenance tasks.

In this scenario, the necessity of context-aware systems arises. Their goal is to offer services adapted to the context of users (e.g. device capabilities, network conditions, user preferences). These systems try to maximize the provisioning of QoS whilst improving the QoE of users, thus, allowing a more efficient usage of resources. In order to provide adapted media communications it is necessary to introduce adaptation and transcoding processes that are transparent for the users. Adaptation mechanisms can be offered by end services or, even, by the network itself. Some advanced examples of these kind of mechanisms are source coding techniques such as Multiple Description Coding (MDC) or Scalable Video Coding (SVC) that allow preparing contents to be transmitted over a network in a scalable and robust manner, adapting communications to network conditions and end device capabilities. Furthermore, there are different quality metrics (objective and subjective) that can be introduced in streaming systems to measure the degree of quality being obtained in an audiovisual transmission. Thanks to these measurements, systems can react when quality diminishes.

Although this PhD. Thesis contributes with specific media streaming solutions, more functionalities to achieve real seamless media communications that meet users' expectations will be required. Even more, there exist lots of existing functionalities that could be combined as desired to achieve specific goals under certain context conditions. In addition, new advances in all fields will introduce new mechanisms that will improve future networks. At this point, the question is: How can we efficiently introduce all these existing and future functionalities (media or not)?

In recent years multiple initiatives have contemplated restructuring the current Internet architecture in order to cope with its limitations. Basically, there are two kinds of architectural approaches aimed to amend current Internet deficiencies. The first ones are evolutionary architectures that propose to incrementally introduce improvements over the current Internet architecture, for instance, by means of creating specific overlays running on top of the current TCP/IP architecture (Figure 1.1). This view fits the vision introduced by content-centric [5] and some information-centric [6][7] architectures.

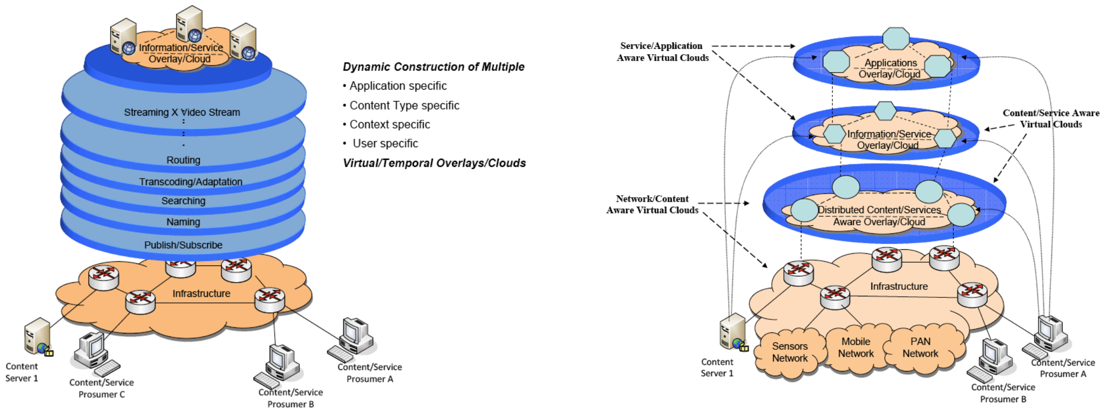


Figure 1.1 Future Internet architecture based on specific overlays [5]

The second ones are revolutionary (also called disruptive or clean-slate) architectures, which intend to introduce new architectures from scratch covering current deficiencies and preparing the Internet constant and fast evolution, motivated by the rapid growth of digital contents, devices, network technologies, services and applications. An example can be seen in Figure 1.2 a) where new information objects can be created and Figure 1.2 b) introducing a completely new architecture not based on layers.

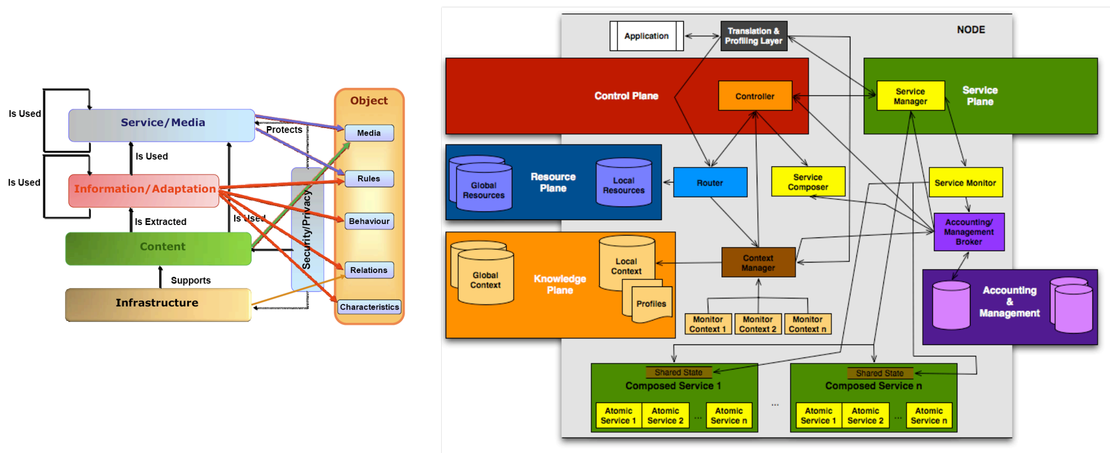


Figure 1.2 a) Mapping a layered-based Content-Centric Internet Architecture into Objects [5], b) TARIFA service-oriented Internet Architecture [8]

Many of the solving strategies introduce novel clean-slate architectures that do not take TCP/IP as their groundwork. Following this trend, Service-Oriented Architectures (SOA) and Service-Oriented Computing (SOC) [58] aim at presenting paradigm principles that can pave the ground to implement a new flexible and scalable architecture for the Future Internet (FI). SOA promotes the usage of services to support the development of rapid, interoperable, evolvable, and massively distributed applications.

Furthermore, in [114] authors presented a very disruptive idea on how to establish network communications, called Role-Based Architecture (RBA). Their proposal was to avoid existing layered structure of the TCP/IP stack and, in so doing, extract the roles or network functionalities which as a result could be interconnected without current layer restrictions (Figure 1.3). Decomposition of current protocols' stack allows a certain granularity of network functionalities that enables their selection as required. This is the first step to provide inherent cross-layering functions to a Future Internet architecture.

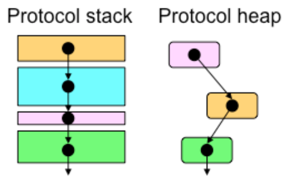


Figure 1.3 Protocol stack vs. protocol heap

Thanks to this approach, numerous specific and complex cross-layer solutions, such as the improvement of switching and routing especially in wireless networks [133], can be avoided. An architecture that does not have layers and does not provide interconnection and hierarchical restrictions could solve the main drawback of these solutions, as is the impossibility to be reused for different purposes in varied situations.

Inter-layer communication in current TCP/IP stack is completely rigid. This rigidity, and its collateral effects, has not only led to different cross-layer approaches, but it has also been a factor in the appearance of sub-layers not considered in the original design, violating the layered structure of the stack [115]. These practices pose serious interoperability issues and provide particular solutions to every problem preventing their reuse. Thus, clean-slate [3] Internet architectures appear in order to propose novel architectures for the Future Internet taking into consideration the limitations of current TCP/IP and the lessons learned from the past. Furthermore, network applications and services continuously evolve, increasing its complexity and requirements, while diverging from the end-to-end principle. New services and computing paradigms require new modes of interaction, new features (identification, context-awareness, seamless service discovery and composition, etc.) and clean solutions to known open issues (mobility, security, flexibility, etc.). For this reason, SOA principles introduce promising guidelines to develop a flexible and scalable architecture from scratch for the Future Internet that will cover current and future requirements, including multimedia communications for a Future Media Internet.

Furthermore, thanks to a context-aware protocol devised to establish communications as well as service negotiation based on specific conditions of the network and the user's requirements, providing services with end-to-end QoS/QoE can likewise be guaranteed. Due to its malleable nature, this will certainly propitiate easy adaptation of Future Internet communications with respect to already well-established requirements and those yet to come.

Instantiating network functionalities only when and where necessary helps to avoid redundancy, which translates into more efficient communications in terms of data throughput and resource usage. Improvements on QoS/QoE provisioning are an immediate consequence as these functionalities can be tracked down and selected as services around the network according to the requirements of the participants involved in the communication and particular context conditions. As an example, energy-aware communications can be provided. Making an efficient use of energy or resources (computational, storage, etc.) is possible. Saving and optimizing resources by providing only minimum (mandatory) network functionalities and resources at each segment of the network would be a feasible fact.

As said, several initiatives have contemplated restructuring the Internet architecture in order to overcome its limitations. However, they all, no matter if disruptive or not, address the issue from network protocol perspectives. The view in this PhD. Thesis is that the Internet has suffered because its original design principles and its requirements cast into common protocols like TCP and, yet today, we see that the emerging networking approaches try to define their own protocol. A fresh new look is required.

This work presents such a fresh new look, away from network protocol and layer rigidities. It adopts a radical view of the Future Internet, where the necessary functionality for accomplishing communications, in user devices and in the network, is not fixed but dynamically composed where and when necessary, as appropriate to user service requirements, network transfer capabilities and surrounding context in the user and the network environments. This work provides a new service-oriented communications framework, enabling adapted communications to actual user needs, in highly heterogeneous environments, seamlessly across different underlying network technologies, IP or non-IP. We are proposing a disruptive network architecture, without being yet another clean slate approach.

Composition of basic network-level services supposes a clean-slate approach to the Internet, while composition of higher level (transport and application) services prompts for an evolutionary approach. Nevertheless, composition of communication services manifests itself as a revolutionary way of looking communications and building communication systems. The main innovations, advancing current state-of-the-art:

- First, a Service-Oriented communications framework, for making up communication functionality as needed, at network, transport and application levels, out of existing functionality. Communication functionality is suitably abstracted to decouple its capabilities from their technology-specific realisation, thus, it is viewed as a service. This allows to incorporate existing functionalities and protocols as well as new functionality in a flexible way and to create "intelligent" networks that can adapt communications to context conditions and users requirements. This is key for providing adapted transparent media services transparently.

- Second, by extending the approach at the connectivity level, a novel service-oriented network architecture for providing end-to-end QoS-aware data transfer services. The paradigm of Information-Centric Networking (ICN) [6] is thus extended to apply for services. Moreover, this architecture is completely aligned with the on-going work within the ISO Future Network working group.

- Third, we introduce specific media services that will allow high quality and seamless content delivery in a transparent manner.

This PhD. Thesis introduces the main concepts in the field of media streaming and Future Internet service-oriented architectures that will guide the readers through the rest of the work in this first part. The highlighted concepts guide the basic research topics and techniques that have been explored to make different contributions in order to advance towards the realization of a more flexible Future Media Internet. The second part of this work describes different streaming scenarios where we have provided specific solutions. Although they are very distinct scenarios, all of them present heterogeneity, network dynamics and high packet loss. These are conditions that this work specifically covers and makes different proposals to mitigate their effects. Then, the third part of this PhD. Thesis introduces the envisaged solutions that will enable a flexible service-oriented Future Internet where functionalities are abstracted as discoverable, composable and adaptive services. Finally, conclusions are inferred in part four and future work is highlighted.

Chapter 2 Multimedia Delivery Challenges for the Future Internet

This section describes the main research challenges to be faced in the Future Internet to obtain seamless multimedia communications [10]:

- **Increasing Bandwidth**. The Internet is growing and advancing in several dimensions: the number of users, the amount and size of the content, the new media and new traffic requirements introduced by new services. Thus, more bandwidth is needed in the end-to-end path and new (e.g. P2P) delivery methods are required.

- **Content adaptation and personalization.** The highly heterogeneous environment in terms of diversity of end user devices, networks and user preferences will remain. To ensure a real seamless access to new media applications, it is desirable that the network itself and the services could automatically realize content adaptation and enrichment inside the network.

- **Content-Centric networks.** Content-aware real-time transmission of future media means that the relative importance of each packet towards increasing the end-to-end utility function is established. That is, the more important packets should be better protected (by allocating appropriately network resources) or should be transmitted first in a scheduling scenario.

- **Content/Information driven routing.** Internet routing system shall be capable to consider associated routing information and metrics for path calculation such as the link quality, security level, energy consumption, priorities or location.

- **New architectures and overlay networks for content distribution.** The main research challenge related to new architecture is the (dynamic, autonomicity and self-organising) creation of overlay network infrastructures to support the provisioning of media services to end-user communities. Some of the issues related to overlay networks have a wider impact and span, in fact, multiple areas. For instance the specification and measurement of QoS parameters and other metrics that can be used to assess the underlying communication technologies achieve network friendliness (via local caches), suggest most suitable service instances for the end-user. A possible approach to deal with these cross-layer issues is to gather information from the underlying networks and combine it with the higher level quality assessment and requirements of applications to adjust the overlay networks.

- **Quality of Experience.** Quantification of QoE using objective and subjective measurements remains a challenging research problem. Furthermore, the relationship between network level QoS measures and overall QoE must be studied.

- **Identity, Trust, Privacy and Security.** The content that is being produced and distributed is increasing rapidly. Users expect to be able to take advantage of the future widespread availability of multimedia content and access to virtual worlds. At the same time, they need to feel confident that their security and privacy is being protected. The increasing complexity and scale of future media systems will make the problems of Identity, Trust, Privacy and Security harder to solve.

- **Content Encoding**. Multi-layered Scalable Video Coding (SVC) offers scalability; Multi-view point Video Coding (MVC) allows for different views of video streaming without drastically increasing the data rate; Multiple Description Coding (MDC) offers an inherited resiliency mechanism with improved PQoS when different sub-streams are received from independent physical or logical paths. However, new media formats and encoding methods (also network coding methods) to offer High Definition (HD) selectable free-viewpoint content coding and delivery, considering the evolution from H.264 2D SVC/MVC to scalable HD 3D, Multiview Video plus Depth (MVD) and selectable free -viewpoint video with interactive virtual panning/zooming. Moreover, new media formats that go beyond video and sound to even other senses e.g. feeling, touching, sensing.

- **In-network content enrichment.** Novel methods for in-network content enrichment and cross-network adaptation will be needed to allow for optimal use of available resources and enriched QoE. By dynamically combining the inherited content scalability (SVC different content layers, MVC different content views and MDC different content descriptions) of the same resource (video stream), transmitted from multiple sources (different servers or peers in case of P2P streaming) and/or received over multiple diverse paths or networks (use the MDC features), on-the fly content adaptation, inherited resiliency and enriched QoE may be achieved. Reconstruction of the content segments may take place either within the network or at the edge of the network (at content aware edge routers) offering transparent streaming to low-end terminals or at the terminal side in case multi-network connectivity is available. Cross-network adaptation and in-network content enrichment especially in P2P overlay topologies, will offer traffic adaptation (load balancing to avoid network flooding), optimal use of available resources (bandwidth), and enriched QoE.

Chapter 3 Future Media Internet Enablers

Today, Internet represents a cornerstone information exchange mean and is the main communication tool not only in the business world. Internet also fosters social interactions. With the proliferation of media capable devices is evolving towards the provisioning of richer and more immersive media experiences. Different stakeholders such as users consuming media services, providers offering advanced media services and operators dealing with an enormous volume of media traffic that needs to be delivered in an efficient manner are constantly introducing new requirements. Moreover, recent improvements in video acquisition and content creation will lead to massive creation of new multimedia contents and Internet media applications, including: live streaming, 3D contents, immersive environments, online gaming, virtual worlds, etc. Thus, the Future Internet will have to deal with a huge amount of media contents and will need to be able to give solutions to its seamless delivery.

The Future Media Internet will not only enable faster ways to go online. It needs to be designed to overcome current Internet deficiencies and to flexibly address emerging requirements and future trends in the area of seamless media delivery such as these ones:

- Media-agnostic network architectures.
- Content-centric networking, including methods for content finding and streaming.
- Support of heterogeneous nodes and devices.
- User-centric/user-generated content provisioning.

In this scenario, research on content enrichment and distributed/community overlay networks (peer-to-peer and cloud) are promising fields expected to bring innovative schemes of cooperation and interaction. In addition, they will open the door to support novel applications, like virtual collaboration, immersive environments, personalised media services (and, in general, any kind of service), virtual groups, network gaming, edutainment, etc. Thus, the interaction with content combined with interactive and multimedia lookup capabilities across distributed repositories, P2P and overlay networks and the dynamic adaptation to the characteristics of diverse devices are expected to contribute with outstanding improvements for the Future Media Internet.

Furthermore, advances in network coding, 3D processing, and dynamic adaptation to the network conditions will foster the appearance of applications such as massive multiplayer mobile games and virtual worlds, whilst introducing new traffic demands and constraints on network architectures.

The Future Media Task Force, European working group proposing solutions for the Future Media Internet [118], states that the Future Media Internet is at the crossroads of digital multimedia content and Internet technologies. Basically, it encompasses two main aspects: media being delivered through Internet networking technologies and media being generated, consumed, shared and experienced on the Internet. The Media Internet is evolving to support novel user experiences that are adaptable to the user, the terminal, the networks and the services, that is, the context. In addition, they stress different strategic research areas including:

- Scalable multimedia compression, transmission, concealment.
- Network coding and efficient streaming.
- Content & context fusion for improved multimedia access.
- 3D content generation leveraging emerging acquisition channels.
- Immersive multimedia experiences.

In this PhD. Thesis specific contributions to go a step forward towards a Media Internet, providing mechanisms to achieve seamless communications for the current Internet, are made. Initially, these contributions were proposed as methods operating at the application layer and working as overlays. However, then, we introduce how these functionalities can be abstracted as services in order to achieve a more flexible Future Media Internet architecture able to adapt itself according to the requirements and context conditions when a communication is requested. This is achieved by adopting service-oriented design principles (see chapters Chapter 5 and Chapter 7). This revolutionary view of the Internet will allow introducing, extending, reusing (combining) all existing services at all levels in a native way, without considering current Internet monolithic design and avoiding complexity (e.g. introduced by particular cross-layer solutions) and unnecessary functional duplicities among layers. From authors' point of view, Future Media Internet requires revolutionary steps. However, considering the worldwide economic crisis, backwards compatibility is vital too. Furthermore, we envisage a progressive migration to the Future Internet where, several networks (based on IP and non-IP networks) will coexist and will be able to interoperate.

Next, we list the main contributions done in this research work classified in different general topics related to media streaming and Service-Oriented Future Internet architectures, key enablers of the Future Media Internet. In addition, the full list of generated publications can be found in APPENDIX VII . Through this document, references to our contributions are made following this reference pattern [GP*].

3.1 Media Streaming Solutions

This section is structured into four main blocks: media distribution, media coding, media adaptation and media signalling. These blocks are required elements that allow providing end-to-end media streaming services efficiently. In this PhD. Thesis, different contributions on these topics have been made.

In general, communication networks supporting media applications are characterised by a wide variability in packet loss, delay, and throughput. Moreover, a variety of receiving devices with different resources and capabilities are commonly connected to a network. In this context, coding and transmission technologies, able to process the content to meet demanding application requirements, are crucial. Thus, mechanisms for scalable media compression, transmission and error concealment, will play key roles.

However, this research goes a step forward and, seeing the difficulties to introduce them in a natural manner due to the rigidity of current Internet architecture, proposes to represent these functionalities as services in order to allow their efficient reuse in a more flexible Future Internet when and where needed by means of service-oriented and clean-slate architectures and context-aware service composition mechanisms.

3.1.1 Media Distribution

Current media delivery systems usually use simple schemes based on sender-receiver or IP Multicast. Both solutions present limitations: the former in scalability and the latter in deployment. A possible solution is to use multicast at application level, which it is known as ALM. There are many techniques that have been developed up to date, but always thinking in current Internet and its constraints. In this thesis, the focus is to distribute high quality media adapted to network conditions and device capabilities. The question is then if current mechanisms are still prepared to work with high quality media and its intensive requirements. The papers where these systems are described do not give much information to respond this question. In fact, the reports always mention performance tests with low rates of transmission, low video quality and an enormous amount of nodes.

This PhD. Thesis contributes with methods for improving the High Quality media delivery in heterogeneous, dynamic and Future Internet environments[1]. As we will see, these techniques are mainly based on Peer-to-Peer delivery methods, Network Coding, Forward Error Correction codes and their mix with scalable and robust video coding techniques to face network and terminal heterogeneity and dynamism with the goal of empowering their benefits. In this thesis we have applied these techniques in different challenging fields such as Peer-to-Peer and Vehicular Ad-hoc Networks. Both scenarios introduce similar challenges in terms

[1] Generated Publications: [GP1] [GP2] [GP3] [GP4] [GP5] [GP6] [GP9] [GP11] [GP12] [GP14] [GP15] [GP17] [GP18]

of packet loss and dynamic channel conditions that make these techniques suitable to be applied in order to obtain robust and reliable data transmissions, especially for video streaming.

Moreover, Network Coding is an emerging paradigm for media and generic data communications in networked environments as the Internet. The main goal behind network coding is the optimization of information flows in a network in terms of throughput and delay maximization. This technique has the potential to significantly increase the network transmission capability, at least, in some scenarios (research on dynamic topologies remains open). Hence, it could be a promising enabler of the Future Media Internet. In current networks such as the Internet, data transmission is achieved through routing. Network coding offers a general approach that assumes, in advanced future media networks, nodes can process and code media streams. Of course, this requires that the nodes inside the network should have computation capabilities. Network coding complements source and channel media coding. While source coding aims at compressing media by exploiting information redundancy, channel coding adds patterns of redundancy into the media stream in order to lower the error rate during transmission. On the other hand, Network Coding allows nodes to perform non-trivial operations on the media stream advancing the node capability from simple routing to “Network Coding”. Thus, Network Coding complements source and channel coding to overcome current practical and theoretical limits of media communication networks. Network Coding aims at adding intelligence and computational power to the nodes. Although the potential of the Network Coding paradigm is huge, there are still many open problems that prevent its successful immediate deployment in real-world applications.

In sections 4.1, 4.2, 4.3.2 and Chapter 6 more details on these topics can be found.

3.1.2 Media Coding

Achieving scalable and robust transmissions requires developing a mechanism that allows a media stream to be split into sub-streams in order to facilitate the delivery of media in diverse network conditions. It is not necessary to receive all substreams in order to carry out the visualization of the transmitted media resource, but if all sub-streams are received, maximum quality can be obtained. For that reason, it is necessary to develop coders and decoders, which must be completely decoupled from the underlying multimedia codec. Some studies indicate that coding systems are more efficient if they are associated to a specific codec. Nevertheless, it can become a limitation, special when a great amount of codecs exist and the system must be able to adapt quickly with the minimum cost. On the other hand, complementary techniques such as Peer-to-Peer and Network Coding delivery methods would improve the robustness of the solution.

In this PhD. Thesis we made contributions[2] in this field by applying media coding techniques such as MDC and SVC (see section 4.3.1 for more detail) in different environments (P2P Internet network and VANETs) and their evaluation by means of specific simulators. Furthermore, a novel adaptive FEC coding of RTP packets has been also explored and proposed in order to transfer video data in a reliable manner (see section 6.3) according to the worst packet loss rate.

These video coding techniques are highly suitable for video processing, storage (SVC) and transmission systems designed to deal with the heterogeneity of current communication networks. Traditionally, providing scalability has been related to significant coding efficiency loss and decoder complexity increase. Primarily due to this reason, the scalable profile of prior international coding standards such as H.262 MPEG-2 Video, H.263, and MPEG-4 Visual has not been widely adopted. Taking into account the past experience with scalable coding tools, the scalable extension of H.264/AVC (SVC) provides much better coding efficiency, high bit rate adaptability, and low decoder complexity. In addition, MDC introduces more flexibility avoiding the hierarchical dependence introduced by SVC. Even, it is possible to mix both techniques, depending on the final application.

Moreover, transmission techniques can benefit from the use of FEC coding for protecting RTP media payloads against packet errors by adding a specified FEC redundant data to the transport stream. Furthermore, the packet level FEC can combined with the interleaving technique to increase the error protection efficiency. This is because the combined schemes can scramble correlated burst packet losses and recover them. Thus, it is necessary to adaptively monitoring the packet loss pattern of wireless channel, in order to apply an effective interleaved FEC protection (see 6.3, [GP4]).

Most communications channels, such as the Internet or wireless links, present wide variations in throughput and packet loss. Bit stream adaptation in such environments is critical in determining the video quality perceived by the end user. MDC and SVC can help to improve the received quality and continuity of the media when losses are present by discarding a number of packets at the source or in the network before reaching the decoder. In so doing, a particular average resolution or bitrate can be achieved. In addition to bitrate adaptation, packets may be lost in the channel. Reasons for this could be because of too much delay, buffer overflow or erroneous packet arrivals at the receiver and, thus, have to be discarded by the receiver or the decoder. A way of dealing with excessive channel losses is to employ error control techniques. However, improved video quality can obtained when a combination of source optimization techniques well integrated with error control techniques are considered together in a cross-layer solution, or even better, in a native framework such as the one introduced in this thesis avoiding complex and particular cross-layer solutions based on current layered TCP/IP architecture.

[2] Generated Publications: [GP2] [GP4] [GP9] [GP11] [GP16] [GP17]

3.1.3 Media Adaptation

Internet heterogeneity is growing in terms of networks, devices and users. The advances of the ubiquitous Internet enabling media anywhere and anytime and the spread of end-user multimedia capable terminals (mobile, wireless and wired) are leading to the proliferation of a wide range of emerging multimedia services. Currently, some of these services have already reached a tremendous market success, such as the case smartphone multimedia applications. This outstanding success has been achieved because a user-centric approach has been followed to design the whole process of content production, content consumption and service management. Indeed, the quality of the user experience, the perceived simplicity of accessing and interacting with systems and services, and the effective and acceptable hiding of the complexity of underlying technologies are determining factors for success or failure of these novel services.

In order to maximize the quality of experience (QoE) of users, multimedia resources must be adapted to the specific context where they are going to be consumed. For this reason, this PhD. thesis contributes[3] to the adoption and integration of different techniques for estimating the level of quality of the received multimedia resources (audio and video) in streaming systems. Specifically, these techniques were used in combination with transcoding and encoding (MDC/SVC) functionalities to meet user requirements.

Subjective and objective quality metrics such as PSNR, SSIM, PEAQ or MOS were studied aiming to measure the perceived quality and to, consequently, adapt systems, for instance when context conditions vary. See section 4.4 for more details.

In the Future Internet, quality assessment tools integrated with services, networks and applications will be necessary to meet consumers and providers expectations whilst making a more efficient use of the available resources and improving revenues.

Moreover, this thesis also contributed in the proposal of novel adaptive mechanisms, not to adapt the media itself, but the communications that transport media. This PhD Thesis made contributions to adaptive systems that can tune media delivery according to specific context parameters such as packet loss rate in order to improve certain processes, for example, to add more redundancy in the network to provide robustness and reliability to packet transmissions using FEC mechanisms or adapting the beaconing rate in a VANET scenario [GP5]. This was achieved by means of specific fuzzy logic algorithms.

3.1.4 Media Signalling

In this PhD. Thesis we used different signaling protocols for enabling Internet seamless multimedia communications, especially for P2P mesh-based

[3] Generated Publications: [GP1] [GP2][GP4][GP5][GP7] [GP15]

distribution systems, P2P multiconferencing systems and SIP-based videoconferencing services[4].

Moreover, SIP-SDPng [119] and MPEG-21 [120] were considered in order to improve the provisioning of services in P2P environments. Generally, current media distribution systems use proprietary signaling mechanisms for registry, set-up and ending a media session among users. Specifically, High Quality systems lack this signaling plane and the configuration must be done manually. It is proposed to incorporate Session Initiation Protocol (SIP) because it offers standard signaling plane, which allows integrating easily in any other SIP standard platform. Besides, integration between SIP and MPEG-21 through SDPng is proposed. This idea is not absolutely novel, but currently a few number of systems support it due to the fact that at this moment this integration is only an IETF proposal (draft) and standardization works continue in development. SIP-SDPng and MPEG-21 promise to solve some aspects associated to media coding information, session mobility, capabilities exchange, etc.

In addition, SIP-SDPng offers the possibility to attach context information, gathered by specific agents running in end-user devices. This feature opens the door for developing new context-aware services in the Future Internet. However, these standards specify very heavy formats, usually XML-based. Thus, it is foreseen that not all of them will be able to run in constrained devices such as sensors (very present in the Future Internet). For this reason, specific ontologies and semantics need to be studied to represent services and resources (such as contents) in an appropriate and light format. This is proposed as future work and it is out of the scope of the current PhD. Thesis.

3.2 Service-Oriented Future Internet Architecture Solutions

Contributions listed here describe how all the media streaming functions that were proposed in the previous section, and generally any functionality, can be integrated into the proposed Future Internet architecture. Considering current Internet limitations, rigid layering design and the challenges that networks must face in the future, techniques for discovering and composing specific functional blocks (Atomic Services) were proposed assuming novel clean-slate architectures based on services. Our main contributions tackle service composition, discovery and context integration. Concretely, the clean-slate architecture proposed in the ambitious TARIFA project [8] was considered as groundwork in this research.

3.2.1 Service Composition

Service composition is a fundamental process that allows combining services once they are discovered. We adopt the following definition for service composition: *composition of those processes or functions required to combine and link existing services (atomic and composed services) to create new processes and*

[4] Generated Publications: [GP1] [GP3] [GP4] [GP15] [GP16] [GP18]

services. Service composition is a main part of Service-Oriented Computing [57], and is a process that plays a key role in Service-Oriented Architectures (SOA [58]).

The Future Internet architecture will have to provide high levels of flexibility and scalability in order to face current Internet limitations. Thus, an architecture where services represent the basic components will be considered, as they can be combined taking into account the specific requirements of a communication. Obviously, this architecture must consider context parameters from different entities: users, nodes, services and networks. In addition, there can be considered other high level requirements such as business goals or socioeconomic aspects when providing a service.

Service composition mechanisms will be key for combining those required services and create the composed services needed at each node along the end-to-end path. Considering the Service-Oriented Architecture proposed in the TARIFA project, a suitable composition mechanism was proposed, selected, validated through simulation and implemented as an architectural component (Service Composer). However it remains as future work the applicability of other selection and composition algorithms (e.g. based on fuzzy logic – section 6.2.5 - for selecting among different compositions, services and mechanisms [GP9][GP5]) in order to compare them and see which one fits better for specific scenarios. Considerations such as response time and performance need to be carefully identified in order to apply the most appropriate one in each situation and depending on the kind of services to be composed (application, transport or network levels). See section 7.4 for further details.

Although this thesis is focused on multimedia services[5], any other service can be composed (e.g. network services such as ACK or FEC functionalities). The service composition process must be generic enough to compose any kind of service.

3.2.2 Context-awareness

Context-awareness is also a key feature for enabling seamless provisioning of multimedia services in the Future Internet. Multimedia services must be able to gather context information from different entities: users, networks, devices, etc. With this information, services can adapt communications (server side or inside the network) in order to maximize the experience of users and, in addition, to improve the utilization of available resources. In context-aware systems, it is especially important to provide mechanisms for gathering and exchanging context data, storage and management context data, processing of context data. Concretely, the management of context information is a very challenging task as this data must be consistent and maintained always up to date if one desires to get the best possible adaptation of a service.

Although this thesis[6] is focused on multimedia services, other services can be represented by means of well-defined interfaces using semantics and ontologies.

[5] Generated Publications: [GP5] [GP6] [GP7] [GP8] [GP13] [GP20] [GP22] [GP23]

In this PhD. Thesis we proposed mechanisms for enabling context-aware communications in IP Multimedia Subsystem (IMS)-based Next Generation Networks (NGN) [121][122][123][124] to improve the enrichment of the services provided by operators. In addition we suggested mechanisms for sharing context parameters during the process of discovering neighbors in a network as explained in [GP5]. In this specific case we propose to use the beacons exchanged in a VANET network to share information about the capabilities of nodes. However, these functionalities can be decoupled from the proposed scenarios and be abstracted as independent services in the Future Internet to be used on demand.

[6] Generated Publications: [GP1] [GP2] [GP5] [GP7] [GP8] [GP13] [GP14] [GP20] [GP21]

Chapter 4 Scalable and Robust Streaming for the Future Internet

In the current Internet the major part of the traffic is video and the most bandwidth-hungry applications are video streaming applications. Specifically, in the audiovisual streaming area, the popularity of P2P real-time and Video on Demand (VoD) streaming applications such as PPLive [11], PPStream [12], UUSee [13], Pando [14], Zattoo [15] has been demonstrated. As an example, PPLive has registered over 110 million users, 2 million users concurrently connected, offers more than 600 channels and has users in more than 200 countries. In addition, Youtube [16], a worldwide well-known web 2.0 streaming applications, is preparing to use P2P computing in order to improve its download rate while reducing transmission costs (especially interesting in current financial crisis time). Thus, Peer-to-peer/overlay streaming prototypes and their implementation are especially interesting for the evolution of current Internet to the Future Media Internet.

Nowadays, P2P supposes a major part of current Internet traffic (50% of Internet traffic in 2008 was consumed by P2P file-sharing applications [17][1]) and keeps growing fast. In addition, regarding to high-consuming P2P video applications, statistics in one of the biggest Chinese Internet Service Providers (ISP) show that PPLive accounts 10% of the total Internet backbone traffic, even more than file-sharing (Bittorrent [18] represents the 8% of the traffic). Some studies [19] manifest that streaming is taking over P2P users for video content. Thus, video and peer-to-peer contents are both rapidly increasing Internet bandwidth demands. Recent reports predict an "exaflood" [20] from advances in video over the Internet, rich media content, and User Generated Content (UGC). Moreover, it is expected that by 2013, the sum of all forms of video (TV, VoD, Internet video, and P2P) will exceed 90% of global consumer IP traffic. In that sense, new systems and studies to optimize future P2P and video traffic may have a very high impact on the future of the Internet. However, for the Future Internet we can propose different types of mechanisms for facilitating the distribution of media. One approach is to create specific overlays, for instance for sending media contents, similar to the over-the-top (OTT) current solutions. In this PhD. Thesis we also propose other more disruptive solutions based on services to provide more clean solutions to, not only video streaming, but also known limitations of the current Internet. As we will see, previous knowledge and existing functions will be reused and reorganized to give solution to them. In section 7.5 and, specially, in Chapter 8 more details are given.

In the field of video transmission, live streaming introduces new challenging problems different to ordinary file sharing. In general, media streaming solutions have different features that determine the operation of the applications. For example, the large volume of media data along with stringent timing constraints, the dynamic and heterogeneous nature of P2P networks and the unpredictable behaviour of peers. These effects can degrade the Quality of Experience of a user and, sometimes, makes the user to leave the platform.

There are several issues in multimedia P2P streaming that must be taken into consideration when facing these kinds of applications. Some of them are the following ones: managing peer dynamicity (churn), peer heterogeneity, efficient overlay network construction, selection of the best peers, monitoring of network conditions, incentives for participating peers and appropriate video coding schemes.

The nature of multimedia content makes it highly sensible to the transmission over networks offering nonguaranteed transmission. Therefore, a reliable multimedia transmission system must involve a reliable video coding scheme. Its use is more than essential. Such a scheme must be flexible enough to meet the P2P network dynamics and its heterogeneity.

According to this, P2P streaming applications should be able to deal well with heterogeneity in order to properly work in different types of networks, support different devices and adapt well to high dynamic environments. In addition it must be able to self-adapt in presence of losses or when context changes (for instance when a user who is watching a TV film in its desktop computer wants to continue watching it in its mobile, this is known as session mobility). To solve that, this thesis considers coding schemes. Concretely, it describes the two major techniques for video coding in this type of P2P environments for distributing media contents: Multiple Description Coding (MDC) and Scalable Video Coding (SVC), also known as Layered Coding (LC).

MDC and SVC are useful in the case of varying bandwidth and losses or erasures due to congestion (e.g. Internet) and unrecoverable errors (such as wireless channels). SVC provides a scalable representation that enhances rate control but it is sensitive to transmission losses. On the other hand, MDC provides increased resilience to packet losses by creating multiple streams that can be decoded independently.

Their features are of special interest, specially focusing on the provisioning of media resources with different requirements among heterogeneous peers and networks, including future next HD content or 3DTV.

P2P media streaming architectures are presented as solutions deployed as overlays over the current IP network in order to enable ALM. P2P technology proposes very interesting solutions in order to obtain scalable, robust and distributed systems.

However, nowadays, video contents are being demanded anywhere and anytime. This PhD also considers emerging and challenging communication environments like VANETs. The challenge of video streaming over VANET can be interpreted by the high channel errors of the vehicles in urban traffic conditions. The high packet loss and limited communication range of the vehicles incur frequent link disconnection and uneven network partition. Many of these effects has been studied also in P2P networks, thus, both environments are suitable for the exploration and proposal of robust and scalable streaming mechanisms.

Media streaming applications can be classified according to the delay tolerance just as shown in [21]. Real-time applications need to have low delay tolerance because of the interaction between the end-to-end users (no longer than a 150ms is accepted [22]) in order to achieve fluid interactivity. However, live broadcast applications typically have no interactivity requirements and, consequently, longer delays are tolerated, commonly up to 30 seconds. This delay cannot be detected without interactivity or without a reference point. In the end, on-demand media applications present greater delay tolerance because the existent interactivity is limited to change the channel or due to VCR-like control.

Video streaming has the following main constraints: scalability, bandwidth constraint, real-time constraint and QoS. These characteristics combined yield a unique application scenario that differs from other typical peer-to-peer applications, including on-demand streaming, audio/video conferencing, and file download (see Table 4-1).

Table 4-1 A Taxonomy of Peer-to-Peer Applications

Category	Bandwidth-sensitive	Delay-sensitive	Scale
File download	No	No	Large
On-demand streaming	Yes	Yes	Large
Audio/video conferencing	Yes / No	Yes	Small
Live broadcast	Yes	Yes	Large

The key problem in a peer-to-peer video broadcast system consists on organizing the peers into an overlay for disseminating a video stream. The main criteria to be considered regarding to overlay construction and maintenance operations can be found in the following list:

- **Overlay efficiency**: The construction of the overlay network must be efficient, because the streaming video requires high bandwidth and low latencies. However, if these applications are not interactive then, a start-up delay can be tolerated.

- **Scalability and Load Balancing**: The overlay must be scalable in order to support a large amount of receptors, and the associated overhead must be reasonable at theses large scales.

- **Self-organization:** The overlay must be built in a distributed manner and it also must be robust enough to support dynamic changes of the peers which take part in the overlay. Moreover, the overlay should continuously adapt to changes in the network, such as bandwidth and latency variances. The system should be self-improving, that is, the overlay should evolve towards a better structure as more information becomes available.

- **Bandwidth constraints**: The system depends on the bandwidth contribution of the present peers so, it is important to assure that the total contribution of the bandwidth of a user must not exceed its access bandwidth capacity.

- **Other system considerations**: Selection of a suitable transport protocol that allows overcoming connectivity restrictions, such as NAT and firewalling traversal.

Other issues to consider implementing a real P2P live streaming are the connectivity between peers and the transport protocols. Further information can be found in [23].

4.1 Types of Overlay

This section describes the most common types of overlays introduced by P2P streaming systems: tree-based and mesh-based. These overlays allow to provide Application Layer Multicast (ALM), as native multicast technology can not be deployed in all the Internet. These techniques can be applied Over-The-Top of current Internet architecture (OTT).

Figure 4.1 shows a taxonomy of different types of ALM and gives some examples of well-known applications using them.

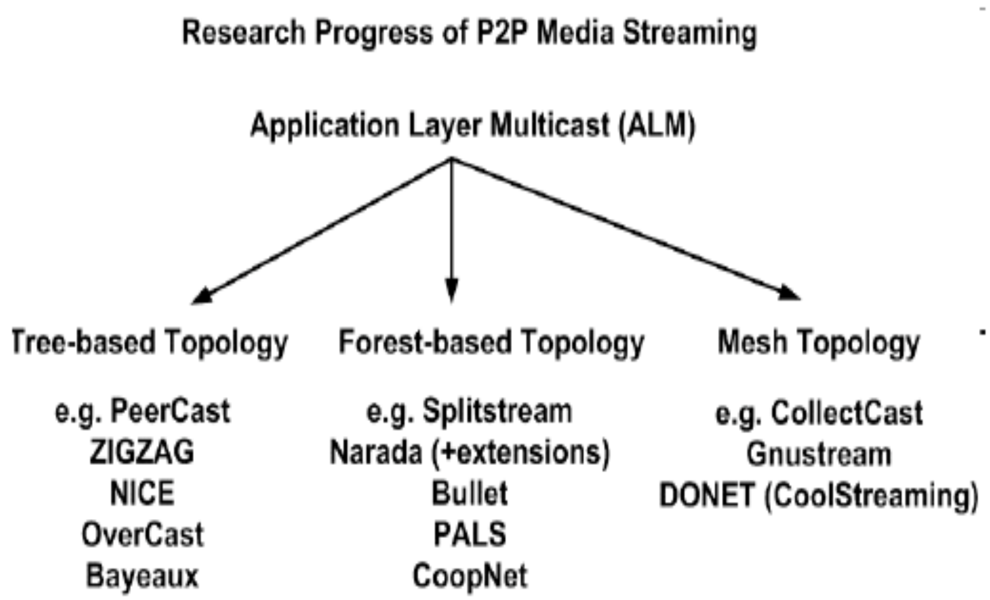


Figure 4.1 Taxonomy of ALM

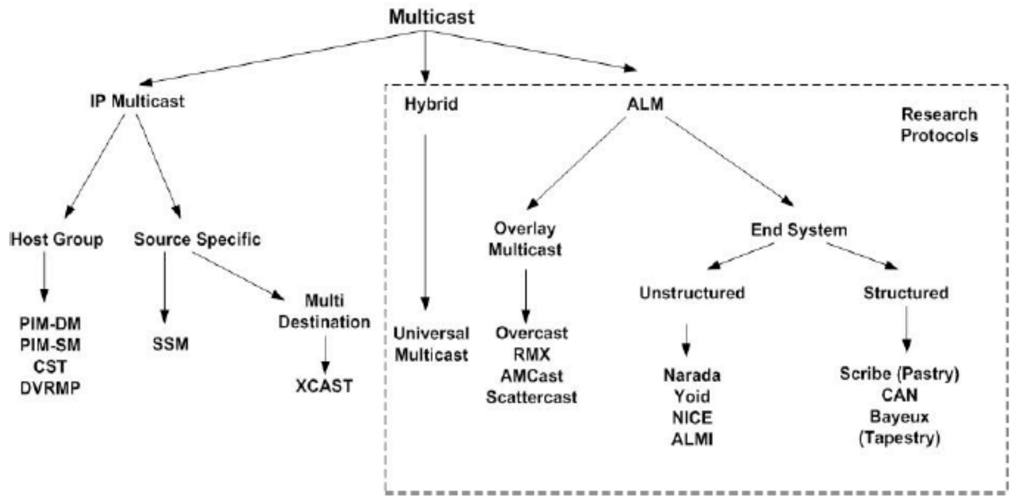


Figure 4.2 Multicast taxonomy

In literature, P2P systems can also be classified into structured and unstructured [24]. This classification is more general and it has relation to the way they organize query handling. Unstructured P2P systems allow nodes to join and leave freely and connect and participating nodes ad hoc into random graph. NARADA [25] and NICE [26] are two examples. Structured P2P systems are designed with the aim of improving the efficiency of data discovery, maintain a logical node structure and impose constraints on the node structure and data placement to ensure effective data discovery. CAN [27], Chord [28], Pastry [29] and Tapestry [30] are some examples. Figure 4.2 depicts a multicast taxonomy where it is possible to see ALM classified in structured and unstructured.

4.1.1 Tree-based Overlay

In tree-based systems [31] the overlay is hierarchically organized. Data is sent from the source node to the rest of nodes of the tree. This approach is known as source-driven. The technique used to send data all along the tree is push-based, that is, when a node receives a data packet, it forwards the packet to each of its children.

There are several requirements for tree-based systems. It is desirable the system to build an efficient tree that matches the underlying network, and also it should provide a swallowed and optimal tree. That tree must be maintainable, that is, if a node leaves or crashes, all the branches of this node will stop receiving packets, in order to repair the tree. Thus, a shallow tree is beneficial.

Finally, cycles are not allowed in a tree-based overlay structure. Moreover, this approach has an important weak point: the failure of nodes, especially when the node is on the top of the tree. The reason is that this “upper-level” node failure may interrupt data delivery to an important part of the tree, momentarily dropping the overall performance. A shallow tree minimizes this problem, but also decreases the upload rate, since most of nodes are leafs and the upload bandwidth of leafs is not used.

4.1.2 Mesh-based Overlay

The construction of mesh-based overlays is an unstructured approach where any explicit data delivery structure is constructed or maintained. Normally, this mesh-based overlay uses a data-driven approach for exchanging data.

Data-driven approach is guided by data availability, which is used to route the data in the overlay. With pull-based techniques (such as CoolStreaming [32]), each node keeps a set of neighbour peers (partners) and periodically exchanges its data availability. Later, one node may retrieve unavailable data from one or more partners and, at the same time, will supply its available data to its partners. Another technique that can be used is the push-pull based approach, such as GridMedia [33], where each node is autonomous and pushes data to its partners without the pull request. The push is performed on time-based prediction, but a wrong prediction leads under flow or duplication issues.

This approach is robust to failures, because available data is redundant (kept in several partners), and the departure of a node simply implies that its partners will use other nodes to receive segments of data (local impact only). The potential bandwidth of partners can be totally used for exchanging data between partners. In mesh-based applications, the scheduling algorithm is a key component, which must schedule the segments that are going to be downloaded from various partners to meet playback deadlines. It is the brain or core component of the system.

Then, an interesting question is: “Is there a future for mesh-based live video streaming?” The positive answer to this question is found in [34] where the authors conclude that mesh-based systems can achieve near-optimal rates in practice with negligible chunk misses (1%) at 90% maximum stream rate, and comparable or better rate than other tree-based systems (AQCS [35], GridMedia).

Also, mesh-based overlay is still an attractive choice thanks to the high rates it offers. It follows a simple unstructured overlay compared with difficult to maintain tree-like structures. It is scalable as delays grow slowly with overlay size and has chunk-tolerance. But the main disadvantage in comparison with tree-based is that the delay remains higher than some tree-based systems.

During the execution of this thesis several contributions have been made to a P2P delivery system named CoolRuc [36][23], which constructs a mesh-based overlay inspired by CoolStreaming.

4.2 Content–aware P2P networks

Among the recent works appeared in the content-aware P2P networks, we could remark the following outstanding approaches.

Active Networks [2] can be viewed as a proposal where data packets contain code fragments with specific processing logic for handling the packet itself. Although Active Networks paradigm introduces a powerful proposal, there are several drawbacks that have hindered their acceptance and deployment. The main concern is security, as there can be active routers running malicious code hidden in active packets. The other primary concern the control of the network. Network operators want to have full control over their network and resources. Next, we make an overview of other content-aware networks.

In DONA (Data-Oriented Network Architecture [37]) users can request named data from the network by using the FIND primitive, while content providers can publish a data object, which will be served to the users by using the REGISTER primitive. To support these two primitives, DONA introduces specific elements called Resolution Handlers (RH), which forward content to the users as an overlay.

Siena (Scalable Internet Event Notification Architectures [38]) proposes a generic scalable publish/subscribe event-notification service similar to DONA. Siena specifies a general model of content-based addressing and routing to maximize both expressiveness and scalability.

PARC [39] started a research program named Assurable Global Networks (AGNs), where they focus on the point-to-multiparty or multiparty-to-multiparty information delivery instead of traditional point-to-point communications. The main feature of AGNs is that the security will reside in the data itself, not in the network as in today pattched Internet. Thus, the network only concerns how to distribute the data and the publishers control the security of the data. Consequently, the network will be a huge storage facility of authenticated content.

OpenCDN [40] constructs an application layer tree using relay nodes that distribute multimedia contents. To coordinate relay nodes, OpenCDN collects client-related information from relay nodes and decides the best relay node for a newly joining client applying specific algorithms.

Slightly different to OpenCDN, Oscar [41] collects sampling information of key distribution during the P2P overlay construction and uses that information to choose routes based on small-world graphs.

COCONET [42] aims to utilize semantic data tagging to provide content level information that will be used by the network for data stream forwarding.

Lastly, Akamai's EdgePlatform [43] is a network of more than 40,000 secure servers with proprietary software, aiming to optimize routes and replicate data dynamically to deliver content and applications more quickly, reliably, and securely. Akamai‘s approach is to eliminate long routes, by replicating and delivering content and applications from servers close to end users.

4.3 Advanced Media Coding Techniques for the Future Internet

During this PhD. Thesis we studied different video coding techniques to prepare the data to be transmitted over the network in order to face the effect of losses, dynamicity of the nodes, dynamic context conditions and network/device heterogeneity. We mainly considered coding techniques for distributing HQ contents over the Future Internet. These coding techniques could be represented as specific functional components (called services) that could be implemented by specific nodes in a network using the Service-Oriented Architecture introduced in (Chapter 7). Then, thanks to the mechanisms described in section 7.3 and 7.4 they will be used to provide adapted communications in a transparent manner for the end users, as they could be discovered and invoked as required although the requester node or provider node does not implement them but a node between them or "in-side" the network.

There are basically two techniques that currently are being studied to be applied to a P2P system able to achieve our goals in terms of scalability and robustness: Source Coding (SC) and Network Coding (NC). Well known examples of source coding techniques are Multiple Description Coding (MDC), Scalable Video Coding (SVC) and Multiview Video Coding (MVC). In general, P2P systems have well known advantages in terms of scalability, robustness and fault-tolerance. However, if all users receive and serve data, the probability that one stream breaks is higher because of the replication rate of the video streams. Furthermore, the connectivity to the network and the different paths used is strongly variable. MDC, SVC and NC are coding techniques that can be applied in media streaming for situations where the quality and availability of connections vary over time.

4.3.1 Source Coding

This section introduces the following main source coding techniques: MDC, SVC and MVC.

4.3.1.1 Multiple Description Coding (MDC) and Scalable Video Coding (SVC)

MDC [44] is a source coding technique, which encodes a signal (audio/video) into a number of N different sub-bitstreams (where N ≥2). Each bitstream is called descriptor (or description), as shown in Figure 4.3 a. The descriptors, which are all independently decodable, are meant to be sent through different network paths in order to reach a destination. The receiver can play the media when any of the descriptors is received.

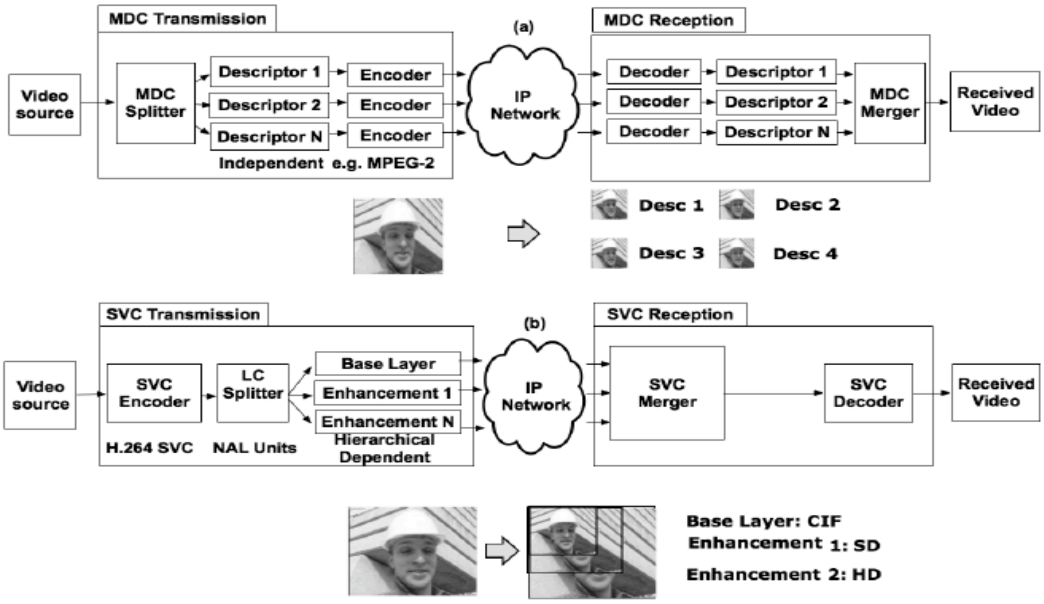


Figure 4.3 a) MDC and, b) SVC main architecture blocks

The quality of the reproduced media is proportional to the number of descriptors received; that is, the more descriptors are received, the better the reconstruction quality is. Since an arbitrary subset of descriptors can be used to decode the original stream, network congestion or packet loss, which are common in best-effort networks such as the Internet, will not interrupt the playback of the stream (continuity) but will only cause a (temporary) loss of quality. SVC [45][46][47], also called Layered Coding (LC), adapts the video information to the network constrains splitting the images into different layers (similar to MDC). These layers represent the quality of the image, so from the base layer each successive layer improves the image quality, getting the full picture quality with the total amount of layers used (see Figure 4.3 b). Specifically, SVC is the name given to an extension of the H.264/MPEG-4 AVC video compression standard. It must be noticed that the main difference between MDC and SVC is that MDC creates independent descriptors (can be balanced or unbalanced) while SVC creates dependent descriptors (unbalanced). According to SVC, we can apply different techniques to make the video data scalable, in the same way as MDC. Currently, real-time coding using SVC technique is still a challenge due to the high computational requirements (to date, there is no SVC real time software encoder, just there are few real-time hardware encoders below HD resolution), which supposes an important limitation at implementation stage.

4.3.1.2 Multiview Video Coding (MVC)

MVC [48] has recently attracted a lot of research. Compressing multi-view sequences independently is not efficient since the redundancy between the closer cameras is not exploited. MPEG and VCEG groups jointly created an ad-hoc group 3DAV [50], which received several contributions for Multi-View coding. A good review on the proposed algorithms can be found in [49]. As an output of this work, Multi-View Video Coding (MVC) is generated as an amendment to H.264/AVC, exploiting temporal and inter-view redundancy by interleaving camera views and coding in a hierarchical manner. The multi-view video codec based on H.264/AVC exploiting the correlation between cameras in a backward compatible way is proposed in [51]. Several prediction structures are proposed with the signalling in the bitstream. Codec is based on baseline profile and using only P pictures. It showed superior performance for dense cameras. First version of MVC extension of H.264/AVC was released in and can be used for some applications such as real time video communication. MVC is one of the first standards towards formal 3D encoding.

4.3.2 Network Coding (NC)

Information theory answers some of the fundamental questions in communications: the ultimate data compression limit and the ultimate transmission rate. One of the most recent research topics in this area is network coding, originally proposed by R.W.Yeung and Z.Zhang in 1999 [125], as an alternative to routing.

Network coding [52] is a mechanism proposed to improve the throughput utilization and, thus, overall network capacity, of a given network topology. The principle behind network coding is to allow intermediate nodes to encode packets. Compared to other traditional approaches (e.g. building multicast trees), NC makes optimal use of the available network resources. Each time a client needs to send a packet to another client, the source client generates and sends a linear combination of all the information available to it. After that, a client receives enough linearly independent combinations of packets and it can reconstruct the original information. In [53], the use of NC to minimize the maximal transmission delay during a multicast session, while maintaining high throughput at the same time, is proposed.

Figure 4.4 shows an example where using NC is also possible to reduce the number of hops and, therefore, the delay in the provided topology. In addition, a benefit of adopting network coding is that no routing algorithms are required. In fact, it is an alternative to routing.

In [52][53], authors introduce the two main benefits of this approach: potential throughput improvements and a high degree of robustness. Robustness translates into loss resilience and facilitates the design of simple distributed algorithms that perform well, even if decisions are based only on partial information.

New coding advances at the application level will strongly influence the design and evolution of future algorithms for communication systems [126]. Specific new families of codes are Digital Fountain codes and Network Coding. These are already influencing the next generation of algorithms for peer-to-peer content distribution.

This approach has been applied in real systems for Linux distributions downloading and for videostreaming software like Avalanche [127][128]. However, it would be necessary to deploy physical devices in the network to support this functionality in an efficient manner. Due to operational costs, this is not an easy task to carry out in a short/mid-term period. In addition, this technology is still in its infancy.

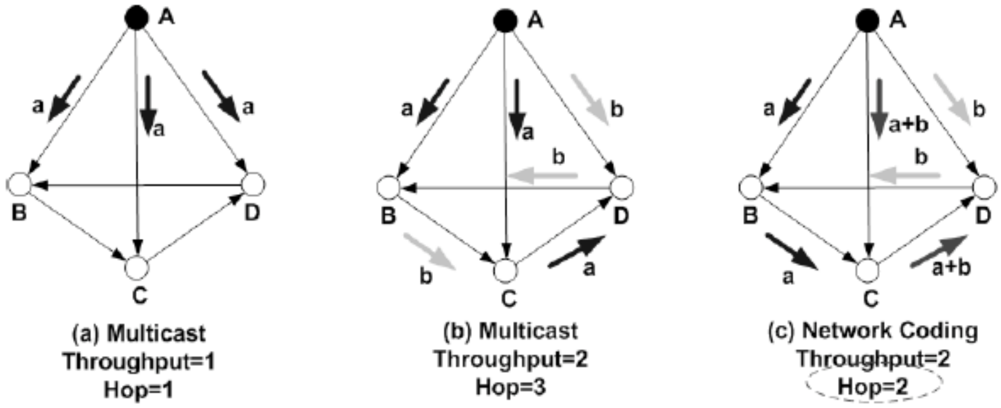


Figure 4.4 Minimizing delay with Network Coding

4.4 QoS Provisioning and Video Adaptation

Interactions across different TCP/IP layers and also among a number of entities along the content or service delivery chain are required to provide services with QoS implies. To achieve an effective adaptation and mapping of QoS parameters at service, application and network levels, cross-layer interactions are usually needed. Moreover, in order to measure and evaluate the QoS of a system, a number of objective and subjective methods have been proposed in the literature. Subjective quality evaluation of contents is a complicated task due to several reasons such as required computation time, cost and human perception. Subjective tests need a large number of tests and trials running under controlled psychometric experimental conditions, to obtain statistically meaningful Mean Opinion Scores (MOS), which allow obtaining the Perceived QoS (PQoS). Obviously, this is not an appropriate solution for real-time audiovisual services. As alternative, objective measurements are used by analyzing the signals in both compressed (e.g. MPEG-4/H.264 compressed video) and non-compressed (e.g. reconstructed RGB video) formats.

As can be seen in Figure 4.5, from the end users perceptual experience to the adaptation final decision, a series of partial QoS can be identified and influence this decision: Subjective QoS, Objective QoS, Network QoS (NQoS), and Adaptation QoS (AQoS). However, more types of QoS (and PQoS) may be defined in an end-to-end environment.

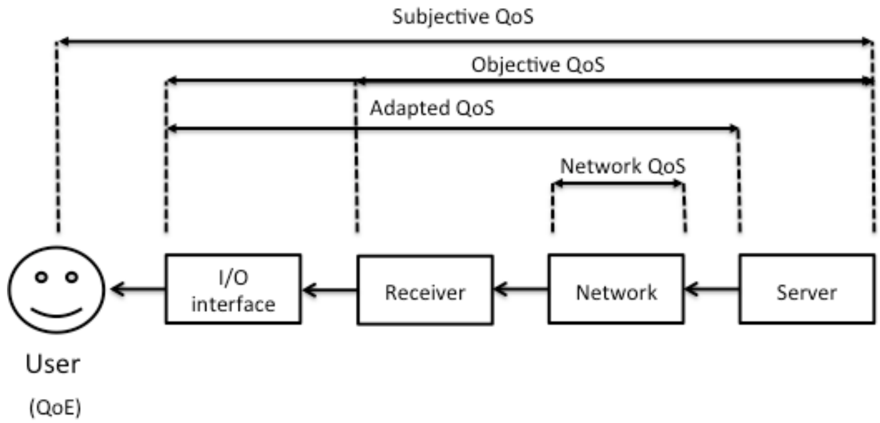


Figure 4.5 Types of QoS in a communications systems

In a layered architecture, such as TCP/IP used in the current Internet, each layer has a set of distinct mechanisms (that need to be optimized to achieve the best possible experience) and related metrics to quantify the behavior of its functionality. Table 4-2 shows examples of some useful parameters and adaptation mechanisms by their respective layers. Parameters presented in this table include both tunable (dynamic) and read-only (static) parameters.

However, it has been shown that adaptation techniques limited to adaptation within a single layer or at application layer (only) are deficient in providing global optimal configurations for the system. In contrast, cross-layer approaches have been extensively discussed in recent research literature for its viability for providing better performance than traditional layered architecture. Although cross-layer design emerged as a by-product of recent proliferation of wireless networks having totally different properties from wired networks, it offers various opportunities for heterogeneous environments, where a variety of application types, network technologies and terminal capabilities are utilized. The main drawback of cross layer approaches is their complexity and the impossibility to reuse them to solve other problems different from the ones they were designed for. This is a fundamental disadvantage that clean-slate architectures for the Future Internet want to solve, introducing new design paradigms such as SOA to provide native-cross layering avoiding complexities and functional duplicities. In this PhD. Thesis we proposed to introduce these techniques to allow adapted media streaming in the Future Internet. See section 8.1 for further detail.

A comprehensive framework that deals with content delivery and adaptation issues is MPEG-21 [54]. MPEG-21 describes a complete content delivery and adaptation framework. Nevertheless, only relatively small portions of the whole MPEG-21 framework have been adopted by industry so far. This ultimately leads to the question whether MPEG has addressed the requirements in a vital way and what needs to be done to foster adoption of the MPEG-21 concepts on a broader scale. New approaches based on SDP signalling have been proposed in [55][56] to provide the required adaptation in the network, without the need for the MPEG-21 signalling and communication overheads. During this PhD. Thesis research stage, contributions on this topic where made (see [GP7][117]).

Table 4-2 Mechanisms and parameters at different layers [10]

Layers	Metrics	Mechanisms (to optimize)
User	Objective (e.g. PSNR, VQM, SSIM) and subjective (e.g. MOS, SAMVIQ) quality metrics	User priority selection, control access
Application	Rate, codec, protection level	Transcoding, FEC, ARQ, adaptive encoding and decoding
Transport	Packet loss, reception window, congestion window, retransmission timer	TCP congestion control, UDP, header compression
Network	IP packet size, DiffServ Code Point	Packetization, Traffic Engineering, DiffServ
Link	Retransmissions attempts, error rate, traffic class, OFDM carriers, TDMA time slot	MAC protocol, Radio resource control, FEC, ARQ, Framing
Physical	Bit Error Rate, signal strength, transmission power, capability profile	Channel modulation and coding
Context data	Battery status, coding profiles, location, bearers, installed applications	Dynamic Voltage scheduling, scheduling, trajectory

Chapter 5 Future Internet from a Service Perspective

The number and diverse nature of Internet services are subject to a speedy increase these days. The range of services currently comprise specific functionalities (e.g. a printing service, video/audio coding function, encryption/decryption function, etc.) as well as complex services (e.g. a holidays booking web page made of a mashup of different web services) which can even be linked to enrich already existing ones. Users want to be able to have access to services anytime, anywhere and no matter how. Specially, this full access to services is a desired feature in the growing new multimedia streaming systems, where users have great quality of experience expectations and, in so doing, introduce really challenging requirements. Gaining access to services is therefore becoming more heterogeneous and dynamic. In this context, an efficient solution to guarantee optimum provision for these services is capital. This solution should take into consideration the requirements of the requester and the context [104] in which they are to be provided. Generally speaking, users hardly ever care about where a service is located. Most users fundamentally contempt to be able to have access to the service under concern in the most efficient way. Internet services are currently accessed via a known IP address or resolved by a DNS service. Once the service has been discovered, it is provided in a best effort manner as Internet does not provide QoS assurance mechanisms. Consequently, it is Future Internet's endeavour to guarantee the best possible user experience by providing an efficient method to discover bespoke services based on specific requirements and context conditions. It must permit to provide adapted services to guarantee the best possible experience of the requester when asking for a service in the network.

Let us not forget that communications depend on the understanding of the involved speakers. They should therefore first of all agree on the terms this communication is to be carried out. Service provisioning rely on a communication that should not only consider technical aspects, but also non-technical issues as well, as the decision sometimes depends on certain subjective parameters or intangibles (e.g. social, economics or regulatory). In fact, users should be able to establish exactly what their target is. It does not only imply to select a final service, but also the characteristics of the service (if it can have different configurations) and, even, switching and routing processes at lower levels. Obviously, many users do not need –may not want or may even ignore- this service granularity when selecting a service, but at least FI architecture should offer the possibility to specify it. This is a loophole that FI should aim at rectifying.

Multihoming, redundancy of services and the paths to reach them are relevant aspects too. Requester should have the whole picture about the services which are available in the network, in so doing be able to establish comparisons and therefore decide which turns out to be more appropriate according to its requirements.

We believe that a new architecture model should be network and technology agnostic. This is a key feature for a better interconnection of heterogeneous networks and devices and should consequently be understood as a crucial consideration for the development of the FI. Such a shift is required in order to deal with the increasing network heterogeneity, including a wide range of services, devices, users and physical technologies. We need to be able to cope with different conditions throughout the complete delivery chain, and thus the intermediate nodes must be aware of the environment and be able to adapt to any sudden change in the network. SOA presents a paradigm suitable for deploying these concepts. Relying on a set of principles for services (loose coupling, abstraction, reusability, autonomy, statelessness, discoverability and composability), architectures based on services are more flexible and able to react more efficiently to changes (context, business, etc.) as compared to the classic monolithic TCP/IP stack. Such flexibility and adaptive capability will be achieved by means of context-aware service composition mechanisms.

Next, Figure 5.1 shows a conceptual service-oriented architecture, highlighting the service composition components that enable creating adapted communications.

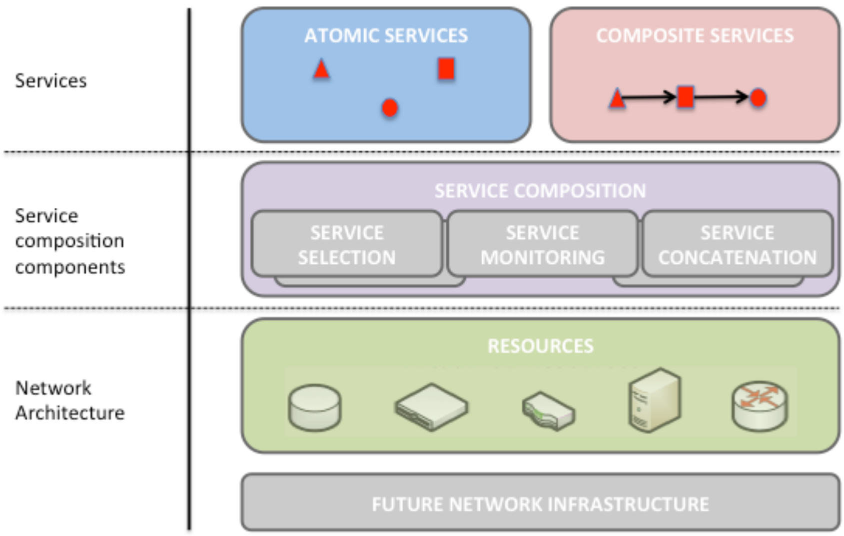


Figure 5.1 Conceptual architecture of a SOA-based architecture

Herein, we think that RBA and SOA offer complementary guidelines for defining a flexible new architecture based on the decomposition of protocols into roles or services and their recomposition according to specific communication goals and conditions. However, the verification of the feasibility of RBA is still an open issue. Whilst it promises higher levels of flexibility, it also introduces new processing requirements and structural changes in comparison to layered approaches. RBA exposes how a packet will be conformed using this strategy and how some roles could be interconnected within a node, but in a real communication it entails other issues that remain unsolved. One of the most important being how different nodes will understand the proposed heaps if they do not follow a hierarchical encapsulation as TCP/IP does.

In order to configure who should execute a role, implementing mechanisms to discover nodes and their functionalities is outstanding especially when considering heterogeneity of interconnected devices. Consequently, a service identification scheme integrated with naming and addressing techniques should be defined to seamlessly provide services. This scheme must be flexible, lightweight and scalable because, unless these properties are met, there is a fair chance to incur in the same problems we encounter these days. Enabling semantic service search will improve the discovery of services, therefore allowing users to seek a service or content based on specific and expressive attributes. For example, using human readable attributes, a user would be able to look for the closest colour printer that has a minimum number of sheets available.

Last but not least, regarding the format of identifiers, we must bear in mind that each router in the core of a network handles hundreds of gigabits of information per second. The built in hardware for routing each packet is dedicated and is responsible for analyzing the packet ID and determining where each packet must be routed. Hardware is restricted by an important limitation. It can efficiently handle size-fixed addresses, while the new addressing scheme must be as flexible as possible. FI routing must take into account these constraints.

Summarizing, an Internet based on services and the proposed premises has some relevant technical challenges [131] [134] that must be tackled such as dynamical service composition of functionalities, context-awareness and automaticity, requester empowerment during service selection and routing, semantic search and service identification. Some of them are faced by the proposed solution, specifically for covering challenging multimedia delivery scenarios.

5.1 Service Definition and Classification

In literature, it is possible to find different definitions for the term "service" (RFC2165-Service Location Protocol, ATIS SON, W3C, OASIS, etc.). In this PhD. Thesis we adopt the following definition:

"A service is any process, function or task that provides networking, computational or informational resources on request which is represented by means of well-defined interfaces"

With this definition we can abstract that a service can be provided by any type of entity/node regardless their characteristics such as hardware, software, virtual, physical, etc. Moreover, a service itself can be of very different natures, ranging from a specific computing function (e.g. audio/video coding or delivery) or storing data to providing resources and tasks to interconnect different entities. Thus, covering from low-level network services like MAC, ACKs and sequencing to end-services or applications like data files (e.g. text, music, video) transfers or web services [129].

Figure 5.2 shows a taxonomy with a possible services classification. This taxonomy is based on [129].

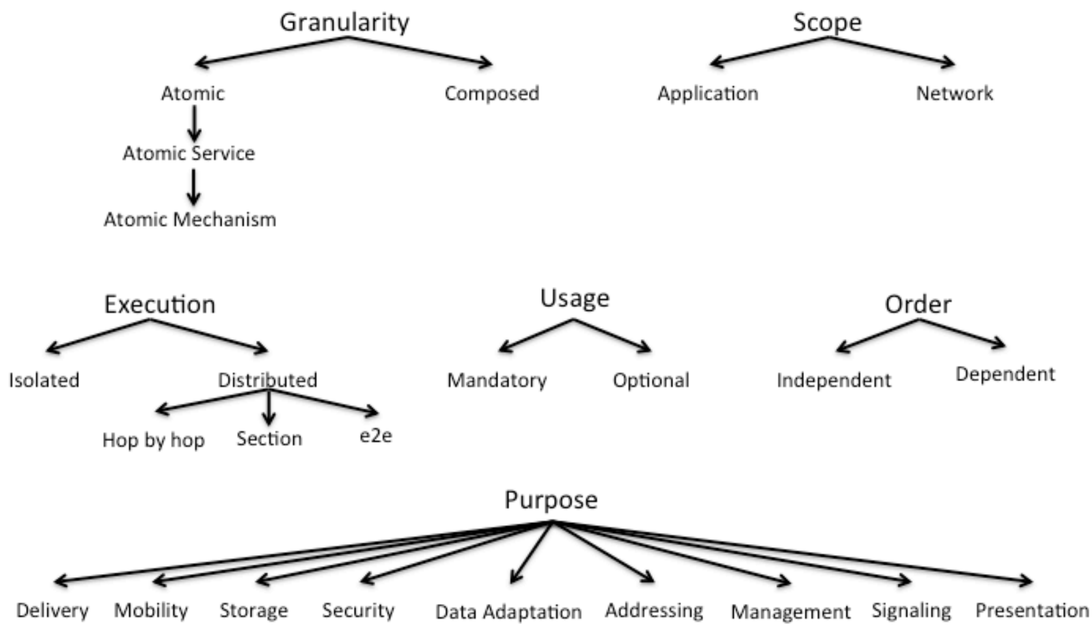


Figure 5.2 Service Taxonomy

Regarding service granularity they can be classified as atomic or composed.

- **Atomic Services** are the core of this architecture. They are the building blocks (roles) used to establish communications and to deliver data in a self-adaptable, self-configurable and context-aware way. Each atomic service provides one concrete and well-defined networking function (along with the reverse function, if any). Different algorithms and implementations of an atomic service may exist (e.g. different congestion control algorithms), and co-exist in the same node, using attributes to both describe the different possibilities and to tune/configure the atomic service in order to use it to fulfill specific workflows needs. As explained in section 7.2, Atomic Services will abstract specific implementations of different functionalities. The specific implementation will be called Atomic Mechanism.

- **Composed Services** are network applications with a wider scope than just establishing communications (e.g. sensing service, directory service, file transfer, instant messaging, presence, etc.). Each composed service or application implies consuming different atomic and sometimes other composed services, with possible dependences appearing between them. In addition, they can involve one or more nodes, depending on the complexity of the service.

- **Execution/Distribution**:

  • **Isolated**: local execution of the service.

  • **Distributed**: execution distributed between two nodes, regardless their location. It includes support for end-to-end, section and hop-by-hop distribution/allocation of services.

- **Scope**: services can be applied with different scopes or considerations depending on the desired result.

  • **Network**: services are executed to optimize communication according network context.

  • **Application**: services are executed to optimize application behaviour and interconnection to meet application requirements according to context characteristics.

- **Usage**: rules governing the service usage.

  • **Mandatory**: usage of this service is mandatory as it is basic for establishing a communication (e.g. forwarding).

  • **Optional**: usage of this service is optional, its usage will depend on application requirements and context characteristics.

- **Purpose**: which is the purpose of the service.

  • **Delivery**: to deliver data between two different entities involved in the delivery chain (they can be adjacent or non-adjacent nodes or they can be two end applications, depending on the scope of the service).

  • **Mobility**: services related to application, user and node mobility.

  • **Storage**: services dealing with the storage of data.

  • **Security**: services dealing with security issues.

  • **Data Adaptation**: services dealing with adapting and transforming data for different objectives, interoperability, customization and optimization of data.

  • **Addressing**: services dealing with the identification and labeling of resources.

  • **Management**: services dealing with the management of the different entities in the network (nodes, applications, services, etc.).

• **Signaling**: services dealing with interchange of signaling and control data.

•**Presentation**: services dealing with the presentation of contents and user/application interfaces.

– **Order**: order/existence of the AS in workflow composition may be dependent of another service.

• **Dependent**: needs the use of another AS.

• **Independent**: no need of other AS execution.

In this PhD. Thesis we proposed some solutions for enabling adapted media streaming communications in the current Internet. All these contributions can be seen as new functional components abstracted as services. Some examples are:

- **Signalling services**: functional components for establishing communications such as SIP-based functionalities (Invite, Register, Info, etc.).
- **Adaptations services**: functional components for adapting video/audio data according to user needs such as coding, decoding, transcoding, quality assessment, etc.
- **Delivery services**: functional components for optimizing the delivery of data in a network such as enabling intermediate nodes to combine incoming data packets (Network coding).

Generalizing this concept, all the existing knowledge (functionalities) could be then abstracted as services and, if present in the network, be used (by means of specific composition mechanisms) to meet user requirements and provide truly QoE to the end users.

5.2 Problem Statement and Requirements of a Service-based Future Internet

A new concept and definition of services for the Future Internet will be closely connected to the innovation of network environments and users. Futuristic capabilities that will be provided by Future Internet and complex and personalized requirements of users can drive that services can be self-evolved continuously. Considering this, the service composition technology should be also extended to cover possible changes derived from the service and network evolution.

New distributed software systems have become more dynamic, allowing transparent distribution, self-reconfiguration, portability, etc. Based on that, new paradigms deviated from the end-to-end principle have emerged, such as Pervasive and Ubiquitous Computing, the Internet of the Things (IoT) or the Internet of Services (IoS).

The continuous evolution of network application and services increased its complexity, adding more and more requirements during the process. New features (identification, context-awareness, seamless service discovery and composition, etc.) are required in order to match with new services and new modes of interaction. Additionally, these features should help to offer clean solutions to known open issues (e.g. mobility, flexibility, security, etc.) [236].

Nowadays, network services are executed without taking into account the characteristics of the surrounding context. Consequently, current architecture is unable to provide information of the underlying network technologies, the capabilities of the devices involved in a communication or the characteristics of the users interacting.

In this scenario, similar or redundant network services (e.g. error correction, retransmission, encryption) are executed at different levels making a bad usage of computing and network resources and sometimes reducing communication performance. Even more, in specific environments, the execution of certain functions can be inefficient for the correct operation of an application or network service. A clear example would be TCP's congestion control in wireless networks specially in media communications. This provokes to modify existing protocols in order to adapt them to environments with certain restrictions. However, a general trend nowadays is to use TCP-based protocols such as HTTP to deliver all kind of data in any environment due to the presence of NAT and Firewalls elements that limit the access of certain protocols to networks. This occurs because of the strongly restricted design of current protocol stack inter-layer communication.

Furthermore, the introduction of strategies in the network that guarantee a certain level of Quality of Service (QoS) and Quality of Experience (QoE) should be as well a critical need in the Future Network (FN).

Taking these problems into account, there are some features that Future Internet should inherently integrate in its architecture. Several challenges must be faced for designing an integral solution for the service composition in FN that allows overcoming the current technologies and deficiencies:

- **Dynamic network composition.** Establishing network functions as services that can provide and access easily to all this information should be an essential feature in FN, in order to have a network that can adapt to current requirements and new ones that eventually rise up. A greater modularity of the network means a better adaptation to new communication paradigms, while decreasing the complexity of the architecture. Unlike static service composition, services can be specified at run time. It means that the capabilities of the service can be extended dynamically, allowing runtime re-composition, decomposition of services, and dynamic adaptation in case of changes in context (services and resources) involved in composite services.

- **Network flexibility.** Network service composition should facilitate in FN the integration of new functionalities into the network. Instead of tight coupling of functionalities within end-to-end protocols, network services are deployed on arbitrary nodes, are loosely coupled and provide their service to arbitrary other functional blocks. This design originates in the service oriented architecture approach and requires means of service description, discovery and composition. This design approach reduces management efforts and provides a flexible framework to integrate new network services.

- **Inherent cross-layer information exchange.** Cross-layer means that functionalities can be adjusted based on the interaction between different layers. FN should allow arbitrary composition and information exchange between network services, thus incorporating the benefits of cross-layer design architectures.

- **Context-awareness in service composition.** FN should support the context management to provide customized and context based services. Thus, different kind of context including user, device, service, resource, and network can be used for discovering, selecting, allocating and composing services to participate in the composition process.

- **Requester empowerment in service choice and routing.** Service requester should have more control over the contents/service that wants to consume. This control must be reflected in flexible routing and service selection according to requester's service definition. Consequently, FN must build a network architecture that provides more intelligence to the network-side whilst still leaving decision-making processes at the end-points.

- **Semantic Searches oriented to service/resource.** FN must be focused on a service/data-centric approach that allows executing the search of services and resources based on the requester requirements. This implies that future network must be able to create, discover, negotiate and consume composite services in a flexible and context-aware way.

- **Resources and services identification.** Every flow over FN must be routed based on its requirements. Therefore, each flow must be identified in order for nodes along the route to cooperate and negotiate autonomously, for guaranteeing the minimum QoS parameters of it.

- **Environmental heterogeneity.** Heterogeneity of nodes, networks and services add another level of complexity to service composition process for FN. If instances of a service are executed in nodes with different capabilities and network access links, every service instance should be evaluated individually, and attributes of a specific one could not be applied to one of another node.

- **Attribute acquisition.** Composition process should be based on the attributes of the services (and their concrete implementation), but extract the complete and updated information of a service is extremely difficult. It should require a previous empiric process extracting information about how the inclusion of a service or another affects in terms of delay, error rate, and each QoS parameter that are relevant for a complete solution (the whole chain of services from requester to end service provider).

- **Service Validation.** Service composition should be validated to guarantee consistency and reliability of services in FN in such a way that it does not hamper the entire process of service composition and heterogeneity of FN. Each service needs to be validated its correctness and consistency before registering itself with FN and composition. Services and composition process must be defined and described with languages based on formal semantics.

- **New Business Models.** Internet brings opportunities to create new business models according to novel services, applications and capabilities demanded by users. Hence, it is necessary to introduce innovative models for costing and pricing. FI architecture should also provide a transparent framework that permits service consumers and providers to interact, establish agreements and satisfy their goals [134]. The new network services can also provide a new business model for network service providers. Nowadays network providers are in strong competition because they all offer the same commodity service - packet transport. Through new network services the network providers can differentiate from competitors and it guarantees higher margins. The network services can be offered as premium services for customers and over-the-top service providers. In this way network providers can also benefit from applications that run over their infrastructure. Because service providers and customers can choose between the services this will also lead to a thriving evolution of network services where the best performing and efficient services will survive [130].

5.3 Relevant Examples of Service Composition in Future Internet Projects and Standardization Activities

Internet faces an architectural crisis. Load on the network is rapidly increasing with the appearance of new users and applications. Since Internet's inception, 40 years ago, lots of patches have appeared aimed to amend the deficiencies of the current Internet. Thus, its architecture is becoming more and more ossified and complex. Some researchers are questioning the principles behind the original design of the Internet that motivates its patch-based evolution. However, nowadays, it remains unclear how the current Internet architecture will be able to cope with all these new requirements.

There are two big approaches in current research trying to solve this doubt. On one hand, evolutionary approaches and, on the other hand, disruptive approaches. From our point of view, it is difficult to believe that evolutionary approaches will allow solving all the issues and challenges that the current Internet introduces (e.g. multihoming, cross-layer interactions, routing table scalability, middle-boxes, QoS, sub-layer proliferation, mobility, security, etc.) in an efficient manner whilst providing enough flexibility to adopt future services and requirements which are yet unknown [115][135][136]. Most of the deficiencies of current Internet derive from the original design principles behind the TCP/IP layered architecture (monolithic, layered and hierarchical stack). The Internet was not designed for its current uses. Consequently, we strongly think it is better to redesign the network architecture from scratch considering current requirements, reusing and adapting all the previous knowledge and advances made during the life of Internet and providing clean solutions to known problems. Therefore we advocate for a clean-slate redesign of the Internet architecture based on service composition approaches. This is part of the work we have been doing in [8][209][GP13].

Recently, there is a big discussion on network architecture design, protocol modularization and redesign of the Internet. The MIT's New Arch project [137] introduced some interesting and relevant work on new network architectures and protocols. The most ground-breaking output of this project was the proposal of the Role-Based Architecture (RBA) [114]. This non-layered and modular architecture build around the concept of roles is one of the bases of this research. RBA establishes that a role can be seen as a communication building block that performs some specific function relevant to forwarding or processing packets. It is remarkable that in this work roles receive can be mapped to the term services. This concept allows establishing relations amongst network components in order to achieve a specific communication goal. Even, we can establish new relationships on demand according to specific needs. In so doing, nodes are able to implement the different roles required for a certain communication. Some examples are packet forwarding, fragmentation, flow rate control, segmentation, request web page, coding/decoding of data or enable/disable caching, etc. RBA proposes the replacement of current protocol stack by a protocol heap, where each node's realm of execution in the protocol heap is limited to its supported roles. Hence, this absence of layered design implies that we need new rules for role combination, modularity, header structure and ordering for the processing of metadata, and encapsulation.

In this work, we adopt an approach based on the micro-modularization of protocol, network and processing functionalities. This is a quite common approach that can be found in other clean-slate proposals [138][139][140][110]. The main idea behind this process is to break existing protocols into small components with a particular functionality. They main difference between them lies in how this modularization is done. For instance, it can be done according a specific goal, scope, focus, purpose and level of granularity. In addition, they also defer in how network architecture and protocols are abstracted, represented and built. Then, communications are composed with these independent components.

Some projects in the USA (NSF GENI/FIND[141][142]), EU (4WARD [139]) and Japan (AKARI Project [143]) have been issued to develop new network solutions from scratch. These clean-slate proposals share some common concepts, like micro-modularization and virtualization as a means to support multiple architectures simultaneously, in their design and objective. Nevertheless, they differ in scope and development. For example, from the GENI/FIND program, the SILOS proposal [138][144] uses micro-modularization similarly to RBA, but does not to avoid creating layers. Actually, they generalize the layering approach to ease cross-layer interactions and avoid sub-layer proliferation. Thus, they advocate building a custom-made protocol stack (known as silo) for each connection made of fine-grain building blocks. This custom stack is the same all across the delivery context, so it is not adapted/tuned to the different requirements of each network section found in heterogeneous environments. Furthermore, it is not clear how silos are negotiated between the edges.

Next, this section makes an overview of some remarkable projects and initiatives that present clean-slate architectures related to SOA, and then an analysis is presented based on the fundamental stages of a service-oriented approach.

The 4WARD project emphasizes the shift from networking nodes to networking information or network of information. This information-centric approach is applied to routing and data transport (as objects). In this approach, protocol design is focused on analyzing protocol invariants in order to build them from their most basic functional blocks (e.g. error control, encryption, etc.). Thus, 4WARD advocates virtualizing the different architectures and protocols using constructions called Netlets made of several functional blocks, in order to meet specific requirements. In this way, nodes instantiate and execute Netlets on top of a kind of protocol virtual machine.

SONATE (Service-Oriented Network Architecture) [110] presents an approach completely based on Service-Oriented Architectures (SOA), considering the Internet as a large, distributed (software) system. Its goal is to encapsulate micro-protocols as services. Thus, their definition of a service relies on open, standardized and generic service interfaces, permitting the decoupling of logic from implementation, and offering an abstract view on the functionality of these protocols, in contrast to hiding mechanisms by layers. However, this specification does not rely on a fixed semantic, to allow later extension to yet unknown practical needs. In addition, context data is not yet considered.

The RINA [145] project proposes a clean slate approach. It proposes a general theory of Inter Process Communication (IPC) where the number of IPC layers (homologous to OSI model) may vary depending on the range of the resource allocation. This network architecture is based on the following principles: (1) Mechanism and policy are separated in the protocols; (2) Connectionless and connection communication modes are unified; and (3) Topological addresses are embedded in a Distributed Inter Process Communication Model.

The Recursive Network Architecture (RNA) [140] explores the relationship between layering, protocols and network architecture. Its primary goal is to encourage cleaner cross-layer interaction, to support dynamic service composition and to gain knowledge on how layering affects architecture. In order to fix the current Internet architecture problems, RNA proposes the use of a generic layer protocol, named metaprotocol. This metaprotocol contains a set of basic services, as well as hooks to configurable capabilities. RNA also notes that the particular services that a protocol provides are dependent on the context (there are environments where a particular mechanism of a protocol is best suited than another one), as well as, on the upper and lower layers that this has.

TARIFA [8] aims at defining a clean-slate approach to a Future Internet architectural redesign, based on a role-based paradigm consisting of non-divisible, or atomic, functions. TARIFA architecture is service-oriented, enabling dynamic composition of services and its adaptation, taking into account context status and its variations. Part of this research was included in the TARIFA project. One goal of TARIFA is that it allows providing adapted communications according to user needs. TARIFA provides a flexible framework of reusable components in order to build context-aware services on top of the architecture. Specifically in TARIFA our contributions were focused on providing a framework for building "custom" context-aware protocols able to react and adapt to changes in context (link conditions, device capabilities, energy constraints, etc.) at run-time. Therefore, services are allocated along the communication path and are composed into a single coordinated workflow for each communication flow (see section 7.4 for more detail). It must be noted that, in contrast with other proposals where the same micro-protocols are allocated at each node, we propose using different functions/modules/micro-protocols at different nodes in order to obtain the desired global behaviour. Another distinct feature of our proposal is the semantic discovery of services, thus incorporating context-awareness in service, and consequently, route discovery. Another difference with other proposals is that we integrate a QoS-friendly protocol for service discovery and negotiation (section 7.3) into the architecture in order to discover services on demand and negotiate and configure the modules to be used by each node in a communication flow.

As seen, relevant Future Internet projects were revised, considering that this research intends to be applied as a part of a foreseen Service Oriented Architecture (SOA) for future networks. Some Future Internet projects take SOA paradigm as the base of their architecture. They define it as services [108], netlets [109] or building blocks [110], but all of them try to tackle the composition of this blocks in more complex workflows in order to provide network and application services. In this context, this section analyzes their approaches in this aspect and its framework similarity to which is considered in this research, in order to understand how relevant could be for our solution the previous work on these projects. Table 5-1 shows a comparison of remarkable international projects from the point of view of the main processes involved in service provisioning. Moreover, APPENDIX II makes an overview on how SONATE and 4WARD carry out service composition.

Table 5-1 International projects comparison

PROJECTS	SERVICE PROVISIONING STAGES				
	Definition	**Publication**	**Discovery**	**Composition**	**Adaptation**
4WARD	Object-oriented, defining Functional Blocks, Functional Block Mechanisms and Netlets as constructions of them. Additionally, NetInf point of view defines Information Objects.	Maintain a repository containing Building Blocks and Design patterns for guiding the composition process. NetInf proposes an approach that uses DHTs to publish Information Objects.	NetInf proposes a dictionary based on DHTs.	Decomposition and recomposition. Chaining Functional Blocks into Netlets. Done at design time by a network or a service architect.	Dynamic behavior provided by a set of policies that tunes the building blocks at run-time In-network management approach considers building block adaptation.
SONATE	Services as essential building blocks that provide self-contained functionality, well-defined interfaces and loose coupled among them. Also defined by a set of effects.	Service Providers publish services at Service Broker.	Service Consumers ask for a service to Service Broker.	Service Broker is the entity in charge of selecting and composing the service matching to the requirements demanded by the Service Consumer. It relies this process on Analytical Hierarchy Process (AHP).	Not considered.
RINA	Services are defined by a set of primitives and arguments, and a set of rules. Associate each protocol to a service definition, but separate its behavior.	Define a Resource Information Exchange Protocol (RIEP) to populate a Resource Information Base (RIB) with application names, addresses, and performance capabilities.	Routing is executed creating the forwarding tables based on RIB information.	RINA bases its composition in an establishment of an Inter Process Communication (IPC) among nodes.	RIEP can be used in both, a request/response mode and a notify/confirm mode, using the same managed objects. This allows IPC processes to notify the other members when there is a

					significant change or to request information.
R N A	Single protocol, (metaprotocol) with a set of basic services (configurable capabilities). Three types of elements: Data element, Control element and Template element.	Not considered.	Not considered.	The metaprotocol is tuned (manual or automatic) depending on the context in which it is located.	Not considered.
T A R I F A	Self-contained, self-describing, modular block offering a specific functionality that can be published, located, and invoked across the network. This self-* definition should permit loose-coupling among services	Distributed Plane (DP) used for maintaining services and context information. DP is divided into a Resource Plane (RP) and a Knowledge Plane (KP).	Propagation of semantic requests specifying the requester requirements. Nodes evaluate incoming requests taking into consideration context information.	Nodes can compose services and adapt them in runtime. Compositions are specified in the form of WFs.	Services can be self adapted considering context variations.

Other less notorious approaches propose interesting points of view about Service-Oriented Architectures for the Future Internet. In [146][147] the concept of service composition is presented, where the Service Overlay Network (SON) is employed as an intermediate layer to facilitate the flexible creation of services and resource provisioning with QoS. Composing basic service components in such a manner allows more flexibility and reusability in building media-oriented services. However, the authors of this work consider that virtualization techniques will evolve and will be a key enabler for the FI. In their attempt to achieve their target, they mention three fundamental issues that need to be solved: (1) service-path calculation issues, (2) resource provisioning problem and (3) the need of a coordination mechanism among distributed services and virtualized resource elements.

In [239], authors propose a programmable network where services cand be integrated inside the network. It introduces how over-the-top services could be allocated inside the network, but a complete framework where users can discover, instantiate and execute services remains undefined.

Closely related approaches are also presented in the last few years in the Software Defined Networking (SDN) field as well. Their spearhead is the

OpenFlow [240] protocol. It defines an open protocol for SDNs that permits an easier creation of services and enhances the flexibility of the network. However, it only applies for core network nodes. Although it is considered a very useful tool for network virtualization, it focuses on the management of a single domain network at layer 2-3 and does not offer a global solution. In addition, it still presents a too static and manual solution. Nowadays, it does not give enough flexibility to empower end-users choice. Dynamic allocation of resources and process delegation to network elements (e.g. a video transcoding process) are uncovered issues. A service-oriented architecture should indeed be required over the OpenFlow substrate.

Substantial prior work has examined the benefits of a new architecture. We view these proposals as complementary to ours. However, this work presents a novel architecture for providing services adapted to context conditions and heterogeneous networks, allowing services to be allocated depending on all the context parameters like links conditions, devices capabilities, users preferences, etc. This is achieved thanks to a generic context-aware service discovery protocol that integrates routing functions.

Some other projects in the USA (CCNx/NDN, Geni), EU (ANA, PSIRP/PURSUIT, SAIL) and Asia (AKARI) have been issued to develop new network solutions from scratch as well. These clean-slate proposals share some common concepts, like micro-modularization and virtualization as a means to support multiple architectures simultaneously, in their design and objective. Nevertheless, they differ in scope and development.

Moreover, there are several initiatives aimed at standardizing service-oriented architectures. ISO/IEC JTC1/SG6 is working on the definition of Future Network considering not only evolutionary approaches but also considering revolutionary or clean-slate ones [148]. ITU-T SG-13 is working on the standardization of Service Integration and Delivery Environments (SIDE) in Next Generation Networks (NGN) [149]. TM Forum [150] is working on several cooperative activities to address the industry challenge of producing an environment for the management of next generation services by means of service-oriented frameworks. It is also working on the standardization of Service Delivery Framework (SDF) for the telecommunication industry. The IEEE 1903 working group focuses on addressing a new paradigm for the Next Generation of IP-based Service Overlay Networks (NGSON) [151] [238]. The Future Internet Assembly (FIA) [152] brings together around 150 research projects that are part of Challenge 1 of the ICT programme of FP7. These projects are advancing the state of the art in the network of the future. Most of these standardization activities contemplate adding new functionalities to the current Internet architecture in an evolutionary way (ITU-T, IEEE, TM Forum). However, revolutionary changes to the stack are also being proposed (ISO/IEC, FIA). Migration of technologies and architectures will be a key issue to guide the evolution of the Internet in order to cover the challenges posed by the Future Internet. This PhD. Thesis is aligned with the work being done within ISO/IEC JTC1/SC6 WG7 "Future Networks"[7].

5.4 Service Identification, Naming and Addressing

An important aspect that the FI needs to deal with is the huge increment of services and, more generally speaking, information available in the Internet in the recent years. Indeed, this situation has indeed impelled several international organizations, such as the IETF, to solve naming, addressing and service identification issues [153]. Most particularly, in the workshop of the Internet Architecture Board (IAB) [154] has raised a number of alternative schemes to address those scenarios by which the current IP-based architecture is at stake and, in doing so, they managed to establish the features to be considered for a new addressing scheme for the FI. Their proposal was a scheme based on the separation of IP addresses into locators and Identifiers. This separation, as proposed in Locator/Identifier Separation Protocol (LISP) [155], would provide the following benefits:

1. Reduction of the routing table size in the Default Free Zone (DFZ).
2. Improvement in cost-effective multihoming in multi-provider scenarios.
3. Easier way of renumbering burden when clients change providers.
4. Traffic Engineering (TE) capabilities.
5. Mobility support without address variations.

The main drawback of LISP is that an appropriate mapping between locators and identifiers must be generated.

In addition to this, recent studies show that LISP adoption brings some important benefits. For instance, [156] shows that the size of the global routing table can be reduced by roughly two orders of magnitude with LISP. This work also shows that LISP provides improved interdomain TE capabilities using an evolutionary approach based on the TCP/IP stack.

Several projects provide solutions for service identification and service discovery processes. A new design called Anycast Name Resolution (ANR) [157] is proposed by the DONA project. DONA aims at replacing both DNS names with flat, self-certifying names, and DNS name resolution with a name-based anycast primitive that is implemented above the IP layer. ANR proposes a separation between the naming and the name resolution. In so doing the naming is responsible for the persistence and authenticity while the name resolution handles availability.

ANR is organised according to *"principals"*. Each *principal* is associated with pairs of public-private keys, and each data item or service or any other named entity (host, domain, etc) is associated with a *principal*. ANR also uses the route-by-name paradigm for name resolution, which is a dynamic mechanism to

[7] The difference between the Future Network (FN) and Future Internet (FI) is that FN covers all kind of networks, while FI can be seen as a subset of the FN.

update routing information in relay nodes. Its operation is similar to the Border Gateway Protocol (BGP) in that it distributes address prefixes between autonomous systems. The route-by-name paradigm distributes the suffixes name availability information among and within the different domains. Furthermore, ANR proposes to add new network entities instead of current DNS servers. These entities are called resolution handlers (RHs). Name resolution is likewise proposed to be undertaken according to two basic primitives: FIND (P:L) and REGISTER (P: L).

Another interesting approach was proposed by The Publish/Subscribe Internet Routing Paradigm (PSIRP) Project [158]. PSIRP proposes a clean-slate architecture for the Future Internet based on the pub/sub communications paradigm, taking nothing (not even IP) for granted. PSIRP borrows basic ideas from DONA proposal.

PSIRP states that all information in the network must be identified. Thus, information is conceived of as the central element (information centric approach) in PSIRP. In addition, the information is hierarchically organised, from small pieces of data to large documents or video files. The way to gain access to information is through information publication. A unique identifier, the rendezvous identifier (RId), identifies each publication. Simultaneously, publications are placed into networks, called scopes (the RIds must be unique within a scope). Scopes can be physical networks, such as a university field, or logical networks, such as social networks. They are organised and each scope has an associated identifier (SId). Moreover, scopes are not only used as mechanisms to locate information, but also as mechanisms to perform access control. The scopes are managed by special elements called rendezvous nodes. In order to get some information published it is therefore necessary to locate a rendezvous node, since it is responsible for managing the scope where this information will be published.

The idea is that during the publication process a user expresses his interest in a specific publication by sending a subscription message from a publication identified by its SId to a rendezvous point (RP). The RP takes full responsibility for the matching between publications and subscriptions as well as the initiation of the publication transfer from the publisher to the subscriber. For a publication to be transferred to a subscriber, a path must be created. In the PSIRP architecture, the function responsible for gathering the topology is responsible for creating the delivering paths in a similar manner to how MPLS creates the forwarding paths. The data forwarding process is performed by allocating a stack of specific identifiers to each publication, called forwarding identifier (FId). INs to send the publication to its destination use the Fid to forward it. When multiple subscribers to a specific Rid occur, a multicast tree is created with the aim of making a better use of the network resources. The publish/subscribe paradigm is used in all levels of operation in the PSIRP architecture, ranging from the internal functions of a node to the transfer of publications over the Internet.

5.4.1 Semantic Service Identification

In the Future Internet users should be able to consume network services anytime, anywhere and anyhow. These are specially needed features to allow a real ubiquitous Future Media Internet. This requirement implies mechanisms to create, discover, negotiate and consume composed services in a flexible and context-aware way.

Service consumers may not know which device provides a desired composed service. Even more, they may not know the name or the identifier of the service they are looking for. However, they know the characteristics of the service they want to consume. Thus, consumers must be able to describe the desired requested service and the network must be able to resolve and reply if there is any service matching this description. In addition, a reachable locator must be provided to access the service. It would be desirable that service consumers could describe a desired service using semantic constructions like "I want a color printer close to building X with toner and paper" and probe the network with this semantic query. As the probe travel the network, traversing nodes match against their profiles if any of their services comply with the semantic service description of the probe. Since each node knows its own capabilities and which composed services it provides and their characteristics, which are described in node and service profile instances, they can match against the attributes of their service profiles if any of their composed services complies with the desired functionality. This process is called semantic service identification. It must be noted that attribute resolution in the semantic identification process is done in-route (see section 7.3 for further details on the proposed semantic requests fields), during packet travelling and not as a previous phase before actually start sending packets.

In order to be feasibly implemented, network nodes must share a common knowledge base or ontology; that is a common attribute semantics and syntax. Although different ontologies may be supported, all nodes must support the minimum identification ontology. This basic ontology is designed to be minimalistic, in order to be supported by all kind of devices and platforms with enough ease (in terms of memory, computing power and energy), but still providing enough level of expressiveness and completeness when building semantic constructions. Attributes are defined for describing node capabilities (CPU, memory, network interfaces, battery, etc.), temporal context characteristics (location, domain), atomic services characteristics (type, supported granularity, dependences, configuration parameters, etc.) and composed services characteristics (I/O behavior, negotiation scheme, description, provider, etc.). As described in [GP13] and [210], in order to minimize the amount of information transferred, attribute syntax could be dictionary-based. Some possible operators that can be applied to attributes when constructing semantic descriptions are logical operators (e.g. AND, OR, NOT, etc.) comparison operators (e.g. <,>, =, etc.) or, even, regular expressions and rules.

5.5 Service Discovery

Distributed computing and resource sharing pose a strong requirement for heterogeneous networks of all types such as large-scale networks, involving different domains (multi-domain) and providers (multi-provider), using different network technologies, while also considering small networks with tiny devices with computation limitations and challenging network conditions. The most obvious repercussion is the difficulty in finding the best services in such a heterogeneous environment according to consumers and the providers' goals, whilst determining the different parties that can participate in a communication.

For instance, imagine a node that wants to establish a communication that involves very different networks, that is, a heterogeneous scenario like the one depicted in Figure 5.3, which involves a segment using mobile networks, optical networks, copper networks, etc. In this environment there may be network segments with reliable communication characteristics. This could be the case for nodes connected with wired links (e.g. segment E in Figure 5.3). Moreover, other network segments could require some mechanisms such as error detection and recovery mechanisms in order to achieve reliability under the effect of high error rates or packet loss in unreliable links (e.g. segment A, B, F). Services should be allocated depending on all the context parameters like links conditions, devices capabilities, users preferences, etc.

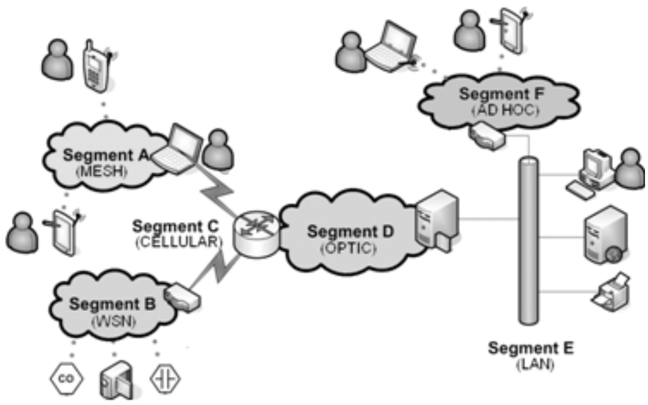


Figure 5.3 Example of a communication involving heterogeneous networks [131]

Hence, in order to provide and consume (composed) networked services in an adapted and context-aware manner (e.g. according to desired functionality or service goal, behavior and QoS constraints), the available atomic services must be suitably combined and allocated along the communication path where they are actually needed and configured accordingly to accomplish the application requirements. Thus, atomic services can be executed in a per-hop or per-section basis.

In the light of this situation, a suitable and intelligent service discovery protocol is the key to find and compose different services in the network. There are diverse Service Discovery Protocols (SDP) proposed in literature. Service discovery enables services and devices to discover, configure and communicate with each other. However, a major problem of current SDPs is that networks with different properties require different discovery protocols and mechanisms. This yields to important interoperability problems. In [159], authors state that existing protocols have various design goals and solutions. Each of them has its own benefits and drawbacks in different scenarios. Therefore, it seems difficult to have a single protocol to discover services in pervasive computing. With current protocols clients and services cannot discover each other if a common protocol is not used.

SDPs have been widely investigated in the field of Web Services and Pervasive Computing. Web Services (WS) are used at the top of the application level and aim towards service globalization in enterprise environments. Enterprise services are customarily consumed within a specific network scope with the presence of firewalls and managed by network administrators. In addition to this, WS scope focuses on application interaction. For instance, WWW Consortium specifies different standards to enable platform interoperability, such as WSDL [160] for XML-based web service descriptions. Regarding service discovery and configuration, we can find three main discovery protocols standardised by OASIS: Universal Description Discovery and Integration (UDDI) protocol, electronic business using XML (ebXML) and WS-Dynamic Discovery (WS-Discovery). UDDI is the oldest, most extended and mature one. The main drawback of WS approaches is that they focus on electronic commerce.

SDPs used in pervasive computing environments [159][161][162][163] are more dynamic and heterogeneous. In [159] an in-depth analysis and comparison of several SDPs commonly used in pervasive computing appear. Some of them are INS, Ninja SDS, DEAP Space, Jini, UPnP, Rendezvous, Salutation, SLP and Bluetooth SDP. This work also offers a useful taxonomy to classify SDPs according to their design principles. In [164], authors offer an accurate comparison of different SDPs (Salutation, SDS, SLP, Jini, INS, UPnP, INS/Twine, JXTA and Splendor) in order to establish grounding guidelines to build a distributed large-scale and multi-domain discovery system capable of facilitating open service discovery across heterogeneous networks and scaling to hundreds of users. They compare them in terms of architecture, effectiveness, fault-tolerance, performance, security, platform and network independence, scalability, interoperability and standardization. Authors focus on three particular approaches: WS, INS/twine and structured P2P systems as potential components of a new SDP considering no one in literature could cover all the objectives defined by them (large-scale multi-domain).

Other works like [40] and [166] contemplate context-aware service discovery. On the other hand, other service discovery systems focus on single hop (Bluetooth [167] and Deep-Space [168]) and multi-hop [169].

Service discovery protocols can be classified in different ways. One classification can be undertaken attending to its centralised or decentralised operation. Another way of classifying SDP is by means of their suitability to be used in different networks (e.g. large vs. small and static vs. mobile), which is actually the approach considered in [170]. Specifically, in the field of pervasive computing, discovery protocols are very diverse in that different networks need different protocols. Pervasive SDPs can be adaptive, meaning that they can be used in different domains [171][172][173]. Another approach is to create a layered structure that allows the coexistence of different legacy SDPs by adopting a service discovery abstraction layer above the others [174]. Finally, it is also possible to implement service discovery gateways to support interoperability between different protocols such as those presented in [175], where Jini clients can interact with UPnP clients and viceversa. In [176], a dynamic service proxy that enables the interoperability of different protocols is introduced. Gateways are considered more efficient than a layered structure. They also make it possible to use legacy protocols.

Regarding Standardization activities, we can point out works done by the IETF for SLP, Globus Alliance for OGSA Grid Services, the W3C and UDDI Consortium (OASIS) for Web Services, Salutation Consortium for Salutation, and UPnP Forum for UPnP. Moreover, SDS, INS and INS/Twine also remain in the stage of research works. A considerable number of existing implementations and development platforms for Salutation, SLP, Jini, INS and INS/Twine, UPnP and JXTA can likewise be found. All these proposed protocols can inspire future service discovery protocol design, like the one offered in this work.

Part II: Scalable and Robust Media Streaming

Chapter 6 Robust and Scalable Streaming in Heterogeneous and Dynamic Scenarios

Recently, the proliferation of multimedia contents and multimedia capable devices has significantly grown and will keep growing very fast in the future according to different studies that have analysed the Internet traffic [1] [4]. Video will be the most present content in the network. This situation has provoked that multimedia should be everywhere and, consequently, depending on the context it is consumed, multimedia streaming systems have to deal with very different requirements and conditions. This section introduces three contributions made in two different scenarios (Vehicular Ad Hoc Network and Internet media streaming using P2P mechanisms), however, all of them introduce advances to allow multimedia delivery in a robust and/or scalable manner in such a heterogeneous and dynamic environments.

The first contribution covers a P2P video streaming scenario. This scenario has attracted the interest of broadcasters, operators and service providers as it presents a reliable solution to stream contents massively in the Internet. Concretely, mesh-pull based P2P systems are the most extended ones. Despite these systems address scalability efficiently, they still present several limitations that difficult them to offer the same user experience in comparison with traditional TV. These ones are mainly the free-riding effect, long start-up delays and the impact of churn and bandwidth heterogeneity. In this PhD. Thesis we studied the performance of Multiple Description Coding (MDC) combined with the use of incentives for P2P-based streaming systems in order to mitigate some of them. The simulation results we gathered show that the use of MDC and incentive-based scheduling strategies improve the overall performance of the system. Moreover, an extended version of the simulator P2PTVSim has been developed to quantify the gain in performance when using MDC and incentives in P2PTV applications. See [GP2], [GP12], [GP14] and [GP15] for more details on these systems and contributions.

A second contribution is made focusing on improving video communication over a Vehicular Ad Hoc Networks (VANETs) scenario. Video communication VANETs has many applications, such as emergency video transmission or inter-vehicle entertainment. However, delivering video to high mobile vehicles faces challenges such as packet loss due to intermittent connectivity and channel variations. Thus, designing a reliable approach for video streaming over such harsh scenarios is needed. Hence, we propose a reliable approach based on random practical network coding and source coding. This approach integrates the benefits of network coding with Multiple Description Coding to achieve robust streaming over VANET. Furthermore, we propose a redundancy controller based on fuzzy inference (implemented using jFuzzyLogic [259]) system to adjust the amount of redundant packets based on vehicular traffic density and SNR of the channel. Simulation shows the proposed approach achieves better protection against packet loss, application layer throughput and video quality. Note that the fuzzy algorithm implemented in this contribution could be used as selection algorithm during the service selection and composition process (see 7.4). However, determining the specific situations where an algorithm behaves better than others remains as future work. This algorithm was implemented and validated in [GP9] and [GP5] in different constrained VANET scenarios. The foundations of them are the same although the final application is different and has been adapted (fuzzy membership functions and rules) according to the final goal.

Finally, a third contribution is made in the field of robust media transmission in a challenging VANET scenario using specific FEC mechanisms. Specifically, in [GP4] we propose an error resilient scheme based on packet level Forward Error Correction (FEC) and interleaving technique for reliable video geocasting over VANET. The proposed scheme is able to adapt channel variations by using Real Time Control Protocol (RTCP) reports of vehicles in the communication range of video source. To achieve a fair comparison with other error protection schemes, we have implemented a proposed error resilient scheme in Network Simulator 2 (NS 2). The results of this simulation-based study show that the proposed scheme is able to improve the perceived video quality and protection efficiency while minimizing bandwidth overhead (introduced by proactive error recovery) in urban vehicular scenarios. Note that this FEC mechanism can be used to protect data transmissions at the packet level. Thus, can be seen as another mechanism complementary to others (e.g. MDC, SVC, etc.) to provide more robust and reliable communications.

Related results and contributions of these works were published in [GP4], [GP5], [GP9], [GP11] and [GP17].

It is important to remark that these contributions propose robust techniques that can be applied to the media delivery (streaming) working at application level, being offered by two completely different applications although using similar techniques to achieve scalability and robustness in different environments. Mainly, these techniques are coding techniques (FEC, MDC, NC). MDC applied in combination with distributed delivery mechanisms (NC and P2P). Moreover, in one hand, the implemented methods could be optimized if they were allowed to get information (e.g. error rate, delay, etc.) directly from the network (lower) layers or specific layers, following a cross-layer approach, for instance, instead of implementing all required mechanisms at the application (highest) layer. On the other hand, applications could share the functionalities they implement among them if they were implemented as services allowing their reuse. To achieve this, this PhD. Thesis proposes to abstract all these basic functionalities as Atomic Services to allow their composition (into more complex Composed Services) according to user and application goals in order to establish and provide adapted services in the Future Internet, specially focusing in multimedia communications. In this case, possible services could be MDC/SVC/MVC functions or operations like NC or FEC calculations like the ones proposed in this thesis. These functions could be natively introduced as services into the network and, in so doing, allow their reuse at any part of the communication just when and where needed. Thus, creating smart networks able to intelligently adapt media according to the specific requirements of users and the different participating stakeholders [237]. This is explained in more detail in section Chapter 7.

6.1 Evaluating Multiple Description Coding with Incentives in P2PTV Systems

P2PTV streaming systems have become a popular service on the Internet (both at commercial and research level), with several successful deployments [177], and the most spread form of what is known as Internet TV. P2PTV allows distributing media contents thanks to multicasting at application level (ALM).

The use of these systems is promising as they offer the possibility to introduce added value to traditional TV broadcasting by providing flexibility, in terms of content delivery (video-on-demand as well as real-time contents), and interactive services. But in order to become a truly successful application they need to be able to provide the same or even better user experience as TV broadcasting offers.

These systems are expected to provide a high degree of scalability with different streaming rates and number of peers. They must also provide continuity under adverse churn conditions (especially in presence of flash-crowds) as well as ensuring delivery of data within a given deadline in order to provide smooth playback. The main aspects affecting the performance of these requirements can be seen in [183]. Among them, we will focus on free-riding (non-cooperation) effect, long start-up delay and the impact of churn and bandwidth heterogeneity in the stability of the system.

In this PhD. Thesis we propose an MDC-based system, which uses incentives for redistribution, in order to address the impact of losses in the Continuity Index (CI) and the delay and the performance problems due to the effect of free-riding. Multiple Description Coding (MDC) [44] is a technique designed to enhance error resilience and increase transmission robustness and scalability. Results obtained in [178] and [179] show that using MDC the delivered quality is acceptable even under the presence of high loss rates. In order to validate the proposed solution we have deployed it in a simulation environment.

For this purpose we are using P2PTVSim (see [180] for more details on this simulator) but with some key modifications. More specifically, we have extended the software in order to be able to simulate the flow of MDC sub-streams in a P2PTV overlay and gather the corresponding statistics as well as the use of a specific incentive strategy. The obtained results show how the use of MDC provides a more robust behaviour against loses (generated by effect of churn and bandwidth heterogeneity). Consequently, the Continuity Index of the system is improved. In addition, thanks to the use of incentives, the effect of free-riding is alleviated.

Several approaches have been studied for the design of P2PTV systems and both simulations and real-environment measures have been performed for the different scenarios considered [177][183]. This work is focused on the use of MDC combined with incentives (inspired by [186]) for redistribution and the goal is to perform the corresponding simulations in order to validate these techniques and discuss their performance.

For the evaluation of the techniques studied in this work we chose to perform a set of simulations instead of carrying out a costly deployment and testing the application in a controlled test-bed. For this purpose, we need a simulator that suits well our requirements and, though the development of a simulator from scratch could be an option, here we have decided on the use of an already developed and tested simulator. The choice has been P2PTVSim which is an event-driven simulator specifically designed for P2PTV systems, that simulates the flow of chunks through a P2P streaming overlay. This simulator has been validated in studies such as [181] and [182]. Other simulators have been considered, but according to the comparison provided in [182] and the characteristics of the simulators, P2PTVSim has been the selected one because it adjusts to the requirements of this study. Some modifications have been introduced in order to enable P2PTVSim for the simulation of MDC with incentives. The result is a variation of P2PTVSim that allows the use of MDC with a configurable number of descriptions as well as the use of the incentive strategy for redistribution.

6.1.1 P2PTV Challenges

P2PTV streaming systems have several design challenges that are crucial for their resulting performance. These applications must ensure an acceptable degree of quality for the received media while being able to manage a large number of nodes that present a highly dynamic and unpredictable behaviour. Also, it is important to point out that delivered data has to meet timing deadlines (related with the playback of the media) in order to ensure proper playback continuity in real-time [189].

As highlighted in section Chapter 4, all these requirements derive in the need to optimize the following metrics: start-up delay (e.g. the delay from when the user selects a content and when the content starts playing), end-to-end delay (e.g. the delay between the content from the source node and the content received at the clients) and the playback continuity, usually measured as the Continuity Index (obtained by calculating the percentage of the overall received data). However, this has to be achieved in a best-effort environment like the Internet where nodes have different bandwidth constraints.

In P2P systems the behaviour of each peer in terms of entering or leaving the overlay is neither coordinated nor controlled. The join and departure events of nodes occur at arbitrary times and these dynamics of peer participation are known as churn and are an inherent property of P2P systems. Churn significantly affects both the design and evaluation of P2P systems. It is important to understand that peers leaving the system during a given interval of time can adversely affect the performance of the system, as some nodes may find themselves disconnected or experience temporary service interruption. According to the results shown in [183] the system performance under conditions of high churn is better for mesh-based systems as they always present better continuity index.

P2P systems rely on voluntary resource contributions by individual peers. The impact of non-cooperation in this type of systems according to [184] is that the level of cooperation affects the overall quality of the system. Higher cooperation is intended to provide better global quality of service and experience. If the level of cooperation is low, the quality of the media is also low even when the network is not congested.

6.1.2 Proposed Solution

This section presents the proposed solution in order to address the issues discussed in the previous sections. Our system is a mesh-pull P2P streaming system that uses MDC with a balanced scheme. It also introduces the use of incentives based on the contribution of the partners in the supplier-side scheduler.

In mesh-pull systems, the video is divided into chunks that are introduced into the overlay by the source and then shared between the different nodes. Each peer exchanges information on the availability of the chunks they have using a representation of their own buffer (known as Buffer Map) and then they share chunks in order to retrieve the video stream within the playback deadline constraints. When the system uses MDC, video is pre-processed before encoding and several descriptions are generated and then encoded and distributed in chunks like common mesh-pull systems do with a single stream. In this case, peers exchange the chunks obtained from each substream (description) and the more chunks they receive from different description the better the received quality will be. Then, an incentive-based mechanism (at the supplier-side scheduler) is used in order to encourage cooperation, so that peers contributing more to their partners are more likely to receive more descriptions and therefore more quality.

6.1.2.1 Receiver-side scheduler

Peers in the overlay request chunks from their partners according to their availability, playback deadlines and other constraints discussed in this section. Each node has a sliding window of chunks that is represented through a Buffer Map to indicate which chunks are already buffered, which ones have been requested and also the chunks that have not yet been requested. Partners exchange their buffer maps periodically and they perform rounds of chunk requests to get the missing chunks in the sliding window considering the availability information provided by their neighbours. The schedule of these requests is critical to achieve an optimal result and retrieve the maximum number of descriptions and ensure the best Continuity Index as well as the best quality.

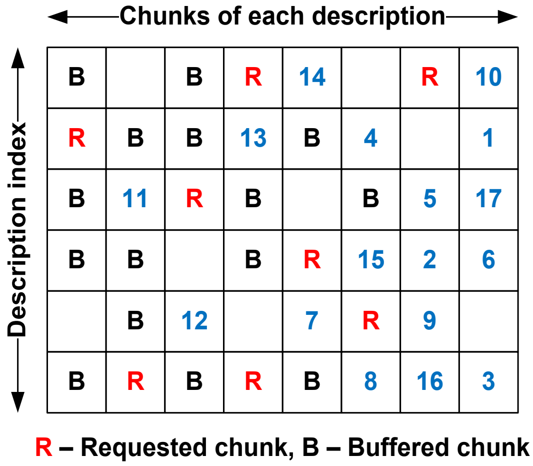


Figure 6.1 Example of Buffer Map for the MDC system

In Figure 6.1 an example of an MDC Buffer Map is shown along with the representation of the chunk requests scheduling order for 17 rounds. Chunks that have already been received are marked with a B and chunk that have been requested with R. The numbers indicate the order of the requests. The considered scheduler looks, at the beginning of each round, for the first chunk index that does not have a buffered (B) or requested (R) chunk from any description. When it finds the first chunk index satisfying this condition it selects a random description (from the available ones according to the availability information provided by its neighbours) and makes the request. Then, it continues with the next chunk index that has none buffered or requested chunk until there are no more chunk indexes fulfilling this condition. The scheduling algorithm continues by doing the same routine over chunk indexes with just one buffered or requested chunk, then with just two, and so on.

The goal of this scheduling algorithm is to get the maximum number of descriptions for each chunk index time but always trying to avoid having high variation between the received number of descriptions from one time to another.

6.1.2.2 Supplier-side incentive-based scheduler

As stated earlier the design considers an incentive-based strategy for the supplier-side that takes into account the contribution of their partners and serves them accordingly. The effect of this supplier strategy is that peers that contribute more are more likely to receive a larger number of descriptions and thus better quality. As a supplier, each peer has a queue of requests from its partners and at each round it has to decide which of these requests is going to be served first. Instead of selecting a random one or handle them in a first-in-first-out manner, a weighted selection is performed. The weight in this case is assigned by computing the percentage of chunks that have been provided by a specific neighbour from the total. Then the selection of the neighbour to be served is done using the weights for each one. Next it is shown the used supplier-side algorithm.

```
Input:
    num_partners :     number of partners the peer has;
    num_descs :        number of MDC descriptors;
    window_size :      size in chunks of the sliding window;
    window :           current buffer map of the peer;
    partners[k] :      buffer map of partner k;
    requested :        set of requested chunks;

Scheduling:
    //look for the 1st chunk index with a number of buffered
    //or requested chunks equal to stage
    stage ← 0
    while no_segment_scheduled and window_not_full do
        i ← 0
        while i < windows_size and no_segment_scheduled do
            count ← 0
            for j to num_descs do
                if window[i , j] ≠ null or requested[i , j] = true
                    count ← count + 1;
                end if;
            end for j;
            if count = stage
                j ← 0;
                //select descriptor to be requested
                while j < num_descs and not_selected
                    if window[i , j] = null
                        chunk ← [i, j];
                        not_selected ← false;
                    end if;
                    j ← j + 1;
                end while;
                //select partner to request chunk
                k ← 0;
                while k < num_partners and no_supplier
                    if partners[k , chunk] ≠ null
                        supplier ← k;
                        no_supplier ← false;
                    end if
                    k ← k + 1;
                end while;
                no_segment_scheduled ← false;
            else
                i ← i + 1;
            end if;
        end while;
        if no_segment_scheduled
            stage ← stage + 1;
            if stage ≥ num_descs
                window_not_full ← false;
            end if;
        end if;
    end while;

Output:
    chunk to be scheduled;
    supplier for required chunk;
```

6.1.3 Simulation Results

Four different simulation scenarios have been considered. The first scenario is the reference system which is a mesh-pull P2P streaming system with single layered video and no incentives (default operation of P2PTVSim). The simulations for these scenarios (Table 6-1) allow us to compare the results for our proposed system with results obtained in the exact same conditions. Then, in a second scenario, we added incentives for redistribution. The third scenario is an MDC based mesh-pull P2P streaming system and the last scenario is a variation of the MDC, adding incentives for redistribution.

Table 6-1 P2P simulated scenarios

Scenario	**Layers**	**Incentives**
I. Reference system	Single-layered (1 layer)	No
II. Reference + Incentives	Single-layered (1 layer)	Yes
III. MDC	Multiple layers (4 layers)	No
IV. MDC + incentives	Multiple layers (4 layers)	Yes

The main parameters for the simulations, that are common for the four different scenarios are the following ones: a total of 1000 peers, with a mean degree of 10 partners and a 5 seconds buffer. Four different types of peers: class A, peers with 5Mbps of upload bandwidth (10% of the total); class B, peers with 1Mbps of upload bandwidth (40% of the total); class C, peers with 500Kbps of upload bandwidth (40% of the total); and class D, peers that act as free-riders, with 0Kbps of upload bandwidth (10% of the total). No download bandwidth constraint is assumed according to [185] and [186]. Simulations last during a distribution of 2500 chunks.

For each scenario several simulations have been performed for different loss values (0%, 1%, 5%, 10%, 15%, 20% and 40%) and each of these simulations has been repeated 10 times and their results have been averaged. The loss model corresponds to a random loss model (Bernoulli).

The metrics that are measured for the evaluation of the techniques are the end-to-end delay, the continuity index (CI) and the average number of received descriptions. The end-to-end delay is measured for all the peers of one class as expressed by (eq. 1).

$$(eq.1)\ Average\ Class\ Delay\ = \sum_{1}^{num\_peers} \frac{\sum_{1}^{num\_chunks}(arrival\ time - init\ time)}{num\_chunks} / num\_peers$$

The continuity index is measured differently for single layered systems (eq. 2) and multiple layered systems. For MDC (eq. 3) it is necessary to consider that only one description is needed for the display of the video (in this work it was considered a spatial MDC technique).

$$(eq.2)\ Continuity\ Index\ (CI) = \frac{Num.of\ received\ chunks}{Total\ number\ of\ chunks}$$

$$(eq.3)\ Continuity\ Index\ MDC\ (CI\_MDC) = \frac{Num.of\ DC\ times\ with\ one\ or\ more\ chunks}{Total\ number\ of\ chunks}$$

6.1.3.1 Reference system

When considering the effect of losses in our reference system we can see that the Continuity Index (CI), decreases as the losses increase. It is important to point out that the degree of 10 partners and the 5 second buffer have an impact in the results and with different (more restrictive) values for these parameters the results would have been worst in terms of CI. In Figure 6.2 the results for the simulations performed for different values of losses are shown.

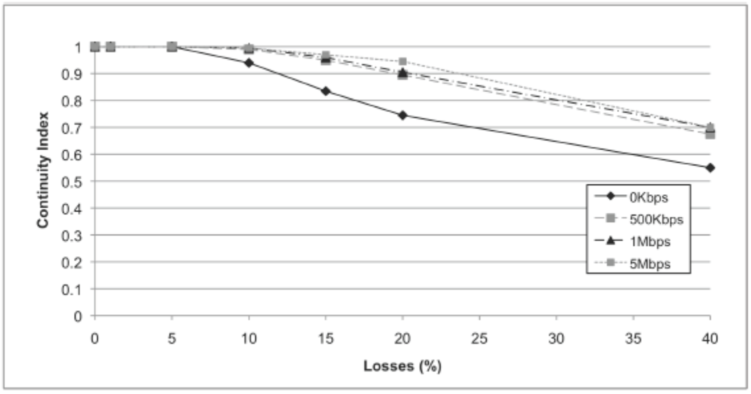


Figure 6.2 Reference system - CI vs. Losses

The delay, as it can be seen in Figure 6.3, increases as the percentage of losses increases and it is in the range of 1,5 to 3,5 seconds, which is coherent considering this delay is the end-to-end delay measured only for the useful received chunks (within the 5 seconds sliding window). Each line has the average values of delay for the peers of the different classes at each loss percentage.

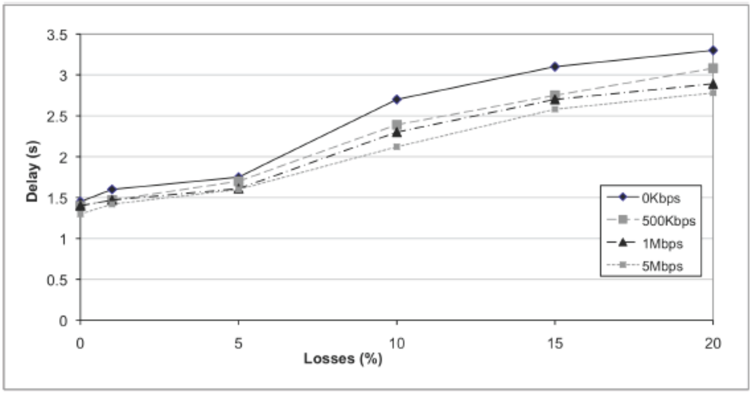


Figure 6.3 Reference System - Delay vs. Losses

The results of the end-to-end delay are also shown in Figure 6.4 but his time depicting the delay for each peer, for three different loss rates (0%, 5% and 10%) instead of showing the average. This result allows us to see the variance of the delay values.

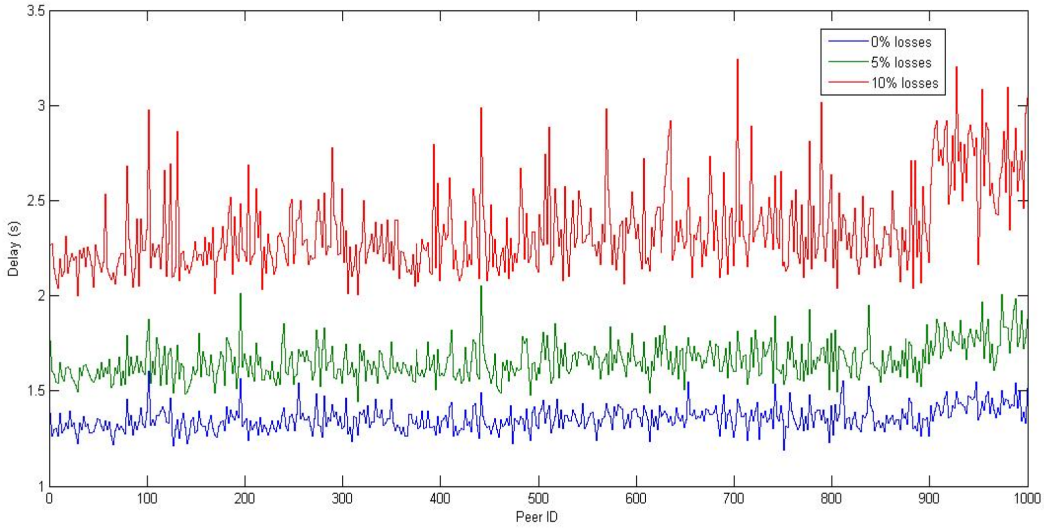


Figure 6.4 Reference System - Delay for 0%, 5% and 10% losses

6.1.3.2 Reference system with incentives

When the use of incentives is introduced as explained in previous sections the overall performance of the system is expected to improve. In Figure 6.5 we can see how the CI of the cooperating peers has increased significantly while the free-riders are aggressively punished having their CI importantly reduced.

Even at high loss rates like 20%, the CI keeps above 0,9 for the cooperating peers and at the 40% loss rate class A peers still maintain the CI over 0,9 while class B and C peers drop to 0,8 and 0,65 approximately. Then evaluating the delay results when using incentives (Figure 6.6) we found that there is also a significant improvement for the cooperating peers, that have a lower delay (being reduced by 1 second approximately). The free-riders delay behaviour as depicted in Figure 6.6, does not effectively drop from the 10% loss rate, but it does drop in the picture because the delay is measured only for the useful received chunks and as it can be seen in Figure 6.5 free-riders do not have that many chunks from that point. Thus, the average decreases as the late received chunks are not included.

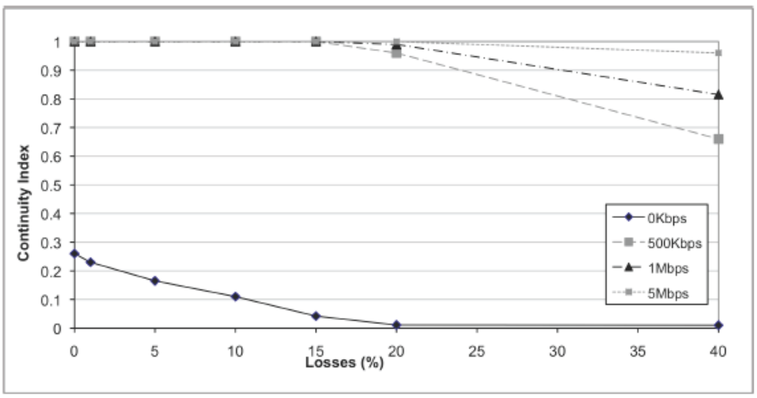


Figure 6.5 Incentive-based system - CI vs. Losses

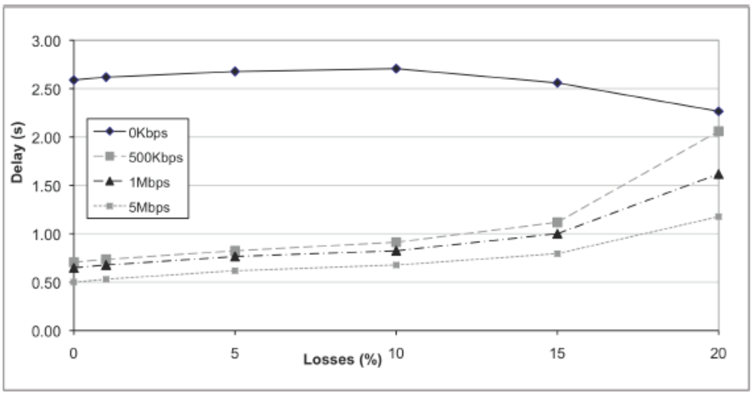


Figure 6.6 Incentive-based system - Delay vs. Losses

The results show that cooperating peers benefit from the incentive-based mechanism. That is because the bandwidth that these peers were wasting serving free-riders is now available and used for these cooperating peers, increasing their overall performance.

6.1.3.3 MDC System

In order to create a more robust streaming system against losses, the MDC scheme is introduced. The proposed system has the same characteristics as the reference one but using multiple layers and the chunk request scheduling strategy (no incentives for redistribution are used). Here the results for the simulations performed with the MDC system are presented. First, in Figure 6.7, we can see the CI achieved using the MDC system. As it can be seen the continuity increases significantly and it only drops to 0,9 approximately when the loss rate is 40%.

This result shows how the use of MDC can improve one of the main metrics that we want to optimize in a P2P streaming system: the continuity playback or CI. But it is important to point out that this measure only indicates the level of continuity (being almost the same for the different peer classes) but does not indicate the received quality. This quality depends on the number of descriptions as well as on the characteristics of the MDC techniques used (such as the ones described in [187]), and to show the difference between the four peer classes the average number of received descriptions is depicted in Figure 6.8. Here we can see that class A peers receive in average a higher number of descriptions than the rest of the classes.

Also, as it can be seen in Figure 6.9, the delay decreases drastically. This is due to the fact that the delay is measured for individual chunks that are smaller in the MDC system (their size in relation to the size of the single layered system chunks can be considered proportional to the number of descriptions used). For that reason the delay here has to be understood in a different context than the delay for the single layered system. Here it is interesting to take into account these low delay values for MDC end-to-end chunk delivery because in systems where there are low-delay requirements MDC could be used.

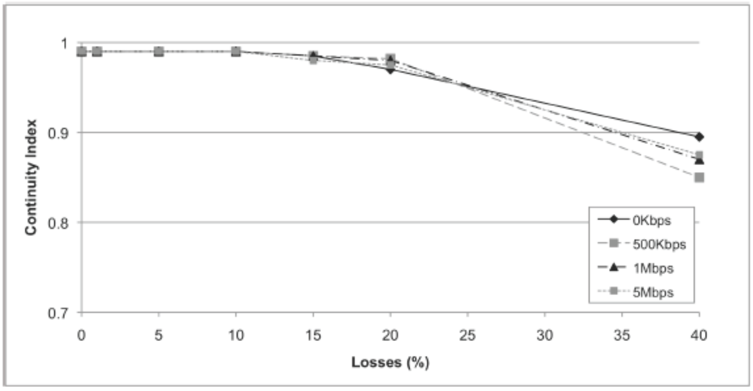


Figure 6.7 MDC system - CI vs. Losses

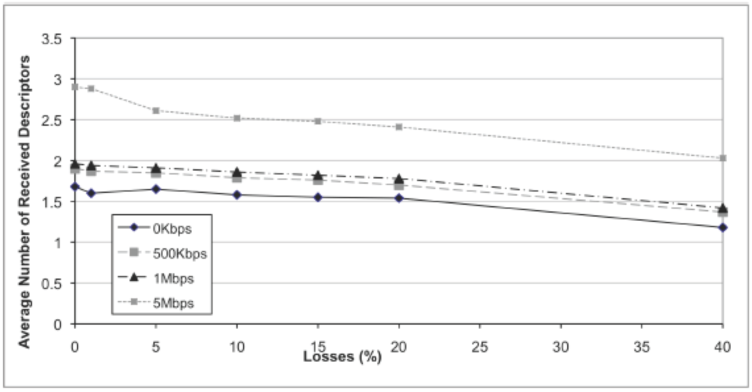


Figure 6.8 MDC system - Average number of descriptors vs. Losses

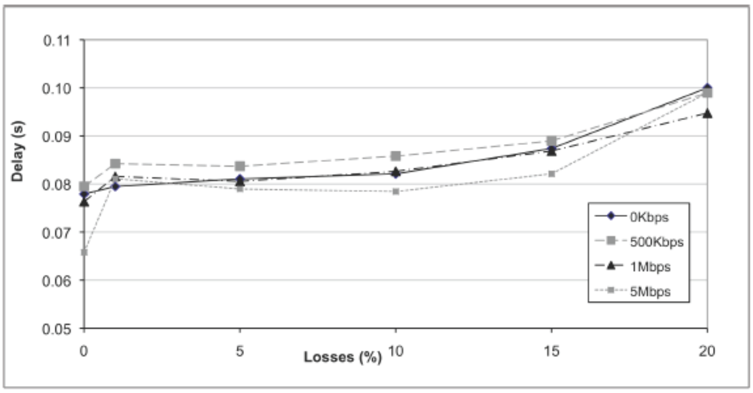


Figure 6.9 MDC system - Delay vs. Losses

6.1.3.4 MDC System with Incentives

Next, we combine the use of MDC with incentives for redistribution. Here, to the improvements introduced by MDC in terms of CI increase and low end-to-end delay, we can add the overall performance boost introduced by the use of incentives. As it is shown in Figure 6.10, the CI can be maintained at almost the maximum level even at high loss rates like 40%. This combination allows a high level of continuity playback. The quality, in terms of average number of descriptions is also increased for the cooperating peers as it can be seen in Figure 6.11. The only metric that is not significantly enhanced in this system is the delay (Figure 6.12) as it is approximately the same delay that the MDC system showed.

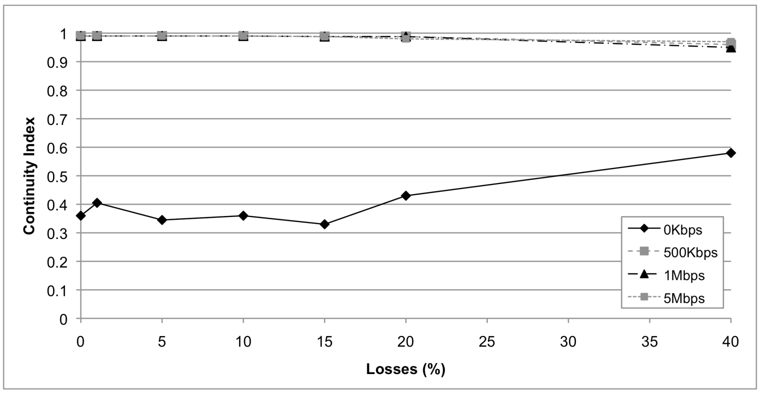


Figure 6.10 MDC + Incentives - CI vs. Losses

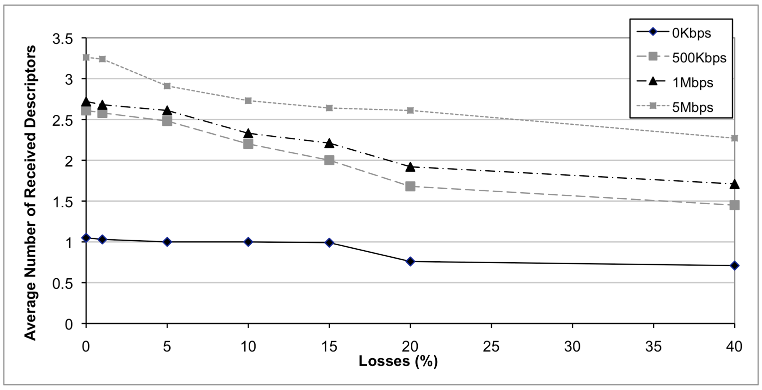


Figure 6.11 MDC + Incentives - Average number of descriptors vs. Losses

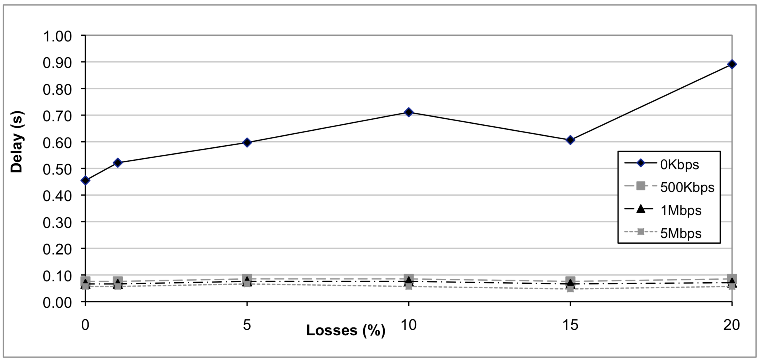


Figure 6.12 MDC + Incentives - Delay vs. Losses

6.1.4 Conclusions

The gathered results show that the proposed solution clearly improves the considered metrics and the overall behaviour of the system compared to the performance of the reference system. These results can be used as reference or guideline for further developments. As an additional outcome of this work we have developed a simulation software which is a valid test-bed that can be used for future studies (based on P2PTVSim). In order to provide a more comprehensive validation of this system other measurements and simulations should be performed to complement the results of this work. More specifically, the measure of the overhead introduced by the use of MDC should be considered in order to see if it is worth deploying this kind of techniques, and trying to define the optimal overhead to be introduced (trade-off between number of descriptors and overhead). Also PSNR quality measurements could be performed by selecting a specific MDC technique and running the corresponding simulations. To complete these results, the effect of churn [190], concretely the impact of flash-crowds, is desired to be added to the simulation environment and evaluated in order to see if the proposed solution performs satisfactorily under these conditions.

Taking into account the obtained results, another line of work that we started was the use of MDC for systems or applications with low star-up delay requirements and the corresponding approaches to it.

6.2 Fuzzy Redundancy Adaptation and Joint Source Network Coding for VANET Video Streaming

Vehicular ad-hoc networks (VANET) are an emerging field of communications technology, which integrates ad hoc network and wireless LAN (WLAN) networks to achieve intelligent vehicle-to-vehicle communication. This kind of inter vehicle communication fosters the deployment of innovative wireless applications based on real time streaming of video flows. Real time video communications between vehicles has several applications ranging from road safety, commercial advertisement and on road entertainment. Furthermore, it is remarkable that the IEEE 802.11 standard specifies the support of video communication over vehicular networks [191].

These networks deal with different traffic scenarios in different situations, such as during late nighttime, rush hour, dense and sparse traffic (e.g. in highways), which can cause unstable vehicular network topologies. Thus, the communication link will last for a short time due to high mobility of vehicles and dynamic topology. Mobility of vehicles leads to high variability of inter-vehicle communication channels according to IEEE 802.11 standard. This makes real time video communications a very challenging task. In addition, the communication channel between vehicles is prone to radio frequency interference together with different forms of fading including multi-path and slow fading. As a consequence, the channel suffers both from high bit error rate and high packet error rate, which leads to high packet loss. There are several reasons for packet loss such as channel error, congestion and packet delay. This work concentrates on packet loss produced by channel error.

In vehicular networks, random and burst errors frequently occur. When bit errors occur during a transmission, the error may repeat over several packets due to the intrinsic dependence among the frames of the compressed video stream. Bursts of errors are considered [192][193][194] to affect more negatively to video quality perception than the effect produced by randomly distributed errors. Thus, to overcome bit error rate, loss recovery methods of video streams are specially needed.

One way to achieve reliable and robust video communications between vehicles is by applying error resilient approaches like Network Coding (NC). In [193] authors showed that NC could provide robustness and high throughput in wired or wireless networks. The main idea behind NC is that intermediate nodes can encode incoming packets instead of having passive nodes just forwarding packets according to routing algorithms as in the current Internet. With NC no routing algorithm is required. The receiver starts decoding data once it has a certain minimum number of coded packets. However, a single packet loss can cause a generation of coded packets to be lost. Therefore, NC should be used with a proper error resilient approach for video streaming in VANET-like scenarios. In addition, NC has the ability to increase robustness against packet loss in wireless ad hoc networks [194]. The error resilient capability of NC depends on the number of encoded packets, which are produced by intermediate nodes. Redundancy may vary according to vehicular network traffic conditions such as sparse and dense scenarios. In sparse scenarios, the number of redundant packets should be increased appropriately to compensate packet loss while in dense scenarios it should be decreased to mitigate bandwidth overload. Thus, we can optimize the redundancy of NC according to the vehicular traffic density in each situation. Also, redundancy can be tuned according to the error rate of the communication channel. In this section, we propose a redundancy controller based on a fuzzy inference system [195], which can effectively adjust the amount of redundant packets to the desired target value.

Furthermore, as seen in section 4.3.1.1, Multiple Description Coding (MDC) encodes a video file into a number of different independent sub-streams. Thus, MDC can be used to reduce the effects of packet loss [44]. Unlike previous works, which do not consider the problems of losing coded packets when using NC, this work proposes to use MDC to mitigate these problems in combination with the adjustment of the NC redundancy according to vehicular traffic conditions and channel error rate variation. Thus, we propose a fuzzy-based NC redundancy control mechanism combined with MDC to achieve reliable video streaming over VANET. This fuzzy-based controller was implemented using the jFuzzyLogic software [259] and has been validated in [GP5] and [GP9] adapting them to the specific goals of each communication. The former for adapting the frequency of the beconing rate in a VANET scenario and the later for deciding the quantity of redundancy data to introduce in a video transmission in a VANET secenario.

6.2.1 Video Transmission over VANET

There has been a few prior works in the field of inter-vehicle video communications. However, almost all of them deal with short message communications between vehicles. Authors in [198] propose an architecture for video streaming over VANET, but no real video data was used in their simulation and only the delay is reported in their experimental studies.

In [199], authors discussed two routing protocols: Source Based Forwarding (SBF) and Receiver Based Forwarding (RBF) applied to VANET networks. Authors considered different traffic conditions for data forwarding to evaluate video streaming between platoons of vehicles.

In addition, video streaming communications over VANET are influenced by the high channel errors of the vehicles in urban traffic environments. The high packet loss and limited communication range of the vehicles incur frequent link disconnection and even network partition. Authors in [201] have combined data mulling techniques with three strategies: network coding [200], erasure coding, and repetition coding. Specially, vehicles in the opposite direction are exploited as data mules to relay multimedia data to other vehicles to overcome intermittent connectivity in sparse vehicular ad hoc networks. However, the analysis of the delay for relaying multimedia data is based on a theoretical mathematical model. Authors in [202] investigated the emergency warning video dissemination to a platoon of vehicles in an accident environment. A network coding algorithm was applied for emergency video dissemination. The performance evaluation reveals that network coding is reliable for video dissemination over VANET especially in high channel loss conditions. Moreover, the authors analytically showed that network coding reduces delivery delay across platoons via data mulling. However, all these approaches did not consider an error resilient scheme to improve the perceived video quality in high channel loss conditions in a vehicular network scenario.

Moreover, in current literature there can be found methods based on fuzzy inference systems. Some authors [199] have used entropy-based fuzzy logic techniques to implement fuzzy controllers with the help of entropy metric-based metrics to reduce the number of route reconstructions whilst being able to provide QoS in ad hoc networks. In [200] authors introduced a fuzzy Petri net agent that is implemented in each node to learn and adjust itself to according dynamic conditions in multicast ad hoc networks. In this section we introduce our fuzzy based redundancy controller, which uses two metrics to adjust the value of redundant packets.

6.2.2 MDC-based Approach

As seen in 4.3.1.1, video coding schemes such as MDC or SVC are used to enhance error resiliency. SVC divides the video file into different layers referred to as base layer and subsequent depending enhancement layers in a hierarchical manner. The base layer is the most important layer while the enhancement layers are referenced to the base layer. The enhancement layers cannot be reproduced independently to the base layer. In contrast to SVC, MDC splits (Figure 6.3) the video signal into multiple sub-streams where each of the sub-streams (descriptors) can be reproduced independently.

Then, thanks to source coding techniques, receivers can make a reconstruction of the video file when any of the sub-streams is received. The quality of the reconstructed video file is proportional to the number of descriptors being received. Thus, the more descriptors are received, the better the perceived video quality. Therefore, MDC is preferable in extreme environments like VANET because it provides increased resilience to packet losses by multiple streams that can be decoded independently. The main parts of MDC system are the splitter and the merger. The splitter is implemented in the source vehicle. The first step of MDC encoder is the pre-processing of the video stream in order to obtain the different N descriptions at the video source, which can be coded and displayed independently. Next, each encoded descriptor is ready to be transmitted.

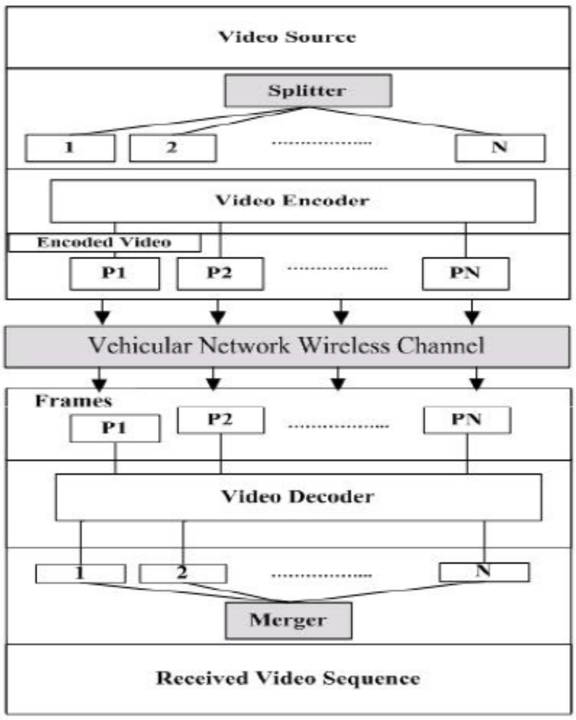


Figure 6.13 System architecture

6.2.3 Random Linear Network Coding Approach

This work proposes to use Random Linear Network Coding (RLNC). When NC is applied in the source node, the video packets are divided into generations with the same decoding deadline [202]. This approach treats each generation as a vector and applies linear combinations before transmitting the data packets. In more detail, let N_1, N_2, N_3,..., N_K be K native packets in a generation. The video source applies linear combinations on these K native packets, and the output is j coded packets which are referred as Y_1, Y_2, Y_3, ..., Y_j , where $Y_j = \sum_{i=1}^{K} e_{i,j} N_{i,j}$ and j = 1, 2, 3,...,h. The set of coefficients $e_{j,i} = \left(e_{j,1}, e_{j,2}, e_{j,3}, \dots, e_{j,K}\right) \in GF(2^q)$ denotes the encoding vector which is randomly selected at the source node to encode the native packets.

Intermediate nodes collect the encoded packets of a specific generation and, then, re-encode the buffered packets. When an intermediate node has received u encoded packets of a given generation, then it will select a new set of coefficients $\varsigma_{f,i} = (\varsigma_{f,1}, \varsigma_{f,2}, \varsigma_{f,3}, \dots, \varsigma_{f,u}) \in GF(2^q)$. Thus, the outgoing packets of the intermediate nodes are X_1, X_2, X_3, …, X_k where $X_K = \sum_{i=1}^{u} \varsigma_{f,i} Y_i$. Note that the new set of coefficients with respect to the native packets is $\gamma_{k,j} = (\gamma_{k,1}, \gamma_{k,2}, \gamma_{k,3}, \dots, \gamma_{k,j})$, where $\gamma_{K,f} = \sum_{i=1}^{u} e_{j,i} \gamma_{f,i}$. Then the intermediate nodes embed the coefficients in the header of the outgoing re-encoded packets to allow their practical decoding. When the destination has received K coded packets, it first checks the linear independence of the coefficients in the header of the received coded packets. Next, it builds K x K matrix using the coded packets. The rows of the resulting matrix represent the coefficients of coded packets. Afterwards, it calculates the inverse of the coefficients matrix and then multiplies it with the received coded packets. A more detailed explanation on how to perform a practical coding and decoding process can be seen in [132].

6.2.4 Proposed Joint Source Coding and Network Coding Approach

Wireless nodes will use 802.11 MAC layer to communicate. In the protocol stack each protocol layer adds all necessary information to the incoming data. In the application layer, specific headers are added whenever needed. Source and destination IP addresses and port numbers are added by network and transport layers respectively. These addresses ensure end-to-end delivery of packets. In case of link disconnection the network layer drops missing packet. In addition, fast channel variations at the physical layer will produce packet loss at the above layers. Thus, in order to increase the robustness of the video streaming system in this VANET scenario, the protocol stack should be modified by embedding data coding approaches to combat packet losses (adopting then a cross layer solution). In this section we propose to achieve this thanks to a joint source coding and NC for improving video streaming in vehicular scenarios.

The overall block diagram of the proposed joint source-network coding is depicted in Figure 6.14. In our video streaming system, a single video source transmits video descriptors to a group of receivers in the VANET network. MDC is applied for robust transmission in this scenario. The MDC generates several descriptors that are independently decodable. These descriptions are passed down to the next protocol layers for performing the end-to-end delivery.

Furthermore, we implemented RLNC mechanisms between network and MAC layers running over an IEEE 802.11 MAC protocol. The reasons for this were:

- First, processing at lower layers (physical and MAC layers) are faster than upper ones.
- Second, to provide transparent NC operations to upper protocol layers.

At the NC shim layer, each description is divided into numbers of generations. This division depends on the size of the generations and MDC descriptions video streams.

The outgoing coded packets are identified by adding a generation-ID and description-ID into its specific headers. This information is useful for the receiver to merge generations and descriptions at the NC and application layers respectively. If a receiver collects enough linearly independent coded packets, the decoder NC component will be able to obtain (recover) the original data belonging to that generation.

Regarding the encoded video, frames depend among them (B and P frames). The loss of one frame may cause other frames to be useless or not decodable. In other words, in certain network conditions a receiver may not be able to decode the original packets that belong to the same generation, because it does not receive enough innovative packets based on its maximum low capacity and, consequently, some frames of a Group of Pictures (GOP) may be lost. If this happens, MDC can compensate the effect of coded packet loss. This is because one description is divided into multiple generations while in NC the effect of one missing generation severely affects the whole video file due to tight correlation between generations to reproduce original video. Therefore, a reliable approach based on MDC and NC is proposed to improve the video streaming service by reducing the multiple packet losses in VANET scenarios. The main drawback is that this method intrinsically adds some redundancy.

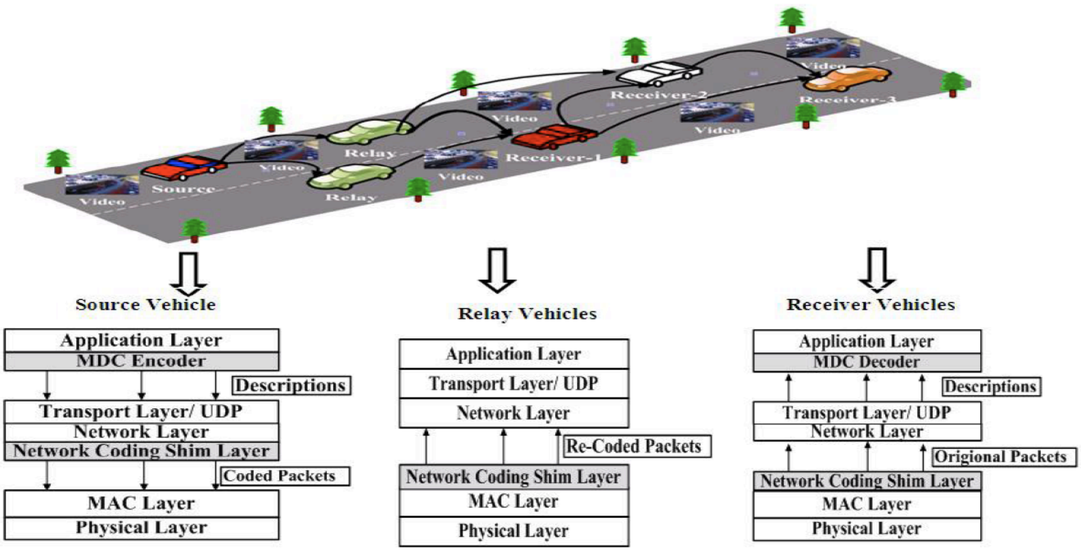


Figure 6.14 Proposed joint source-network coding (dark blocks)

6.2.5 Proposed Fuzzy Logic Redundancy Control

Fuzzy logic simulates the interpretation of uncertain sensed information, as a human brain would do. The appropriate decision making process of fuzzy inference systems depends on the precise design of membership functions and fuzzy inference rules. However, in VANET the right decision-making process is difficult to obtain due to uncertain/random movement of vehicles and the variability of the SNR value of the communication channel. Artificial intelligence based decision making systems, such as fuzzy logic, perform well in pattern classification and decision making systems. Fuzzy logic is a decision making process based on input membership functions and a group of fuzzy rules. This is similar to the way the human brain operates, which simulates the interpretation of uncertain sensory information. Therefore, the fuzzy inference system (Figure 6.15) has been applied to control the amount of redundant packets for improving network load. Concretely, the first step when designing a fuzzy controller is to determine the impact of each input/output parameter and the range of values that each variable can take. The next step is to determine the membership functions for the input and output parameters based on the defined range and to design rules for the fuzzy inference system. One main characteristic of the fuzzy controller is that it is modifiable. It is easy to tune rules (add, modify, delete), membership functions, range of values or, even, change the system parameters to enhance its performance. The selection of the most relevant input variables is crucial.

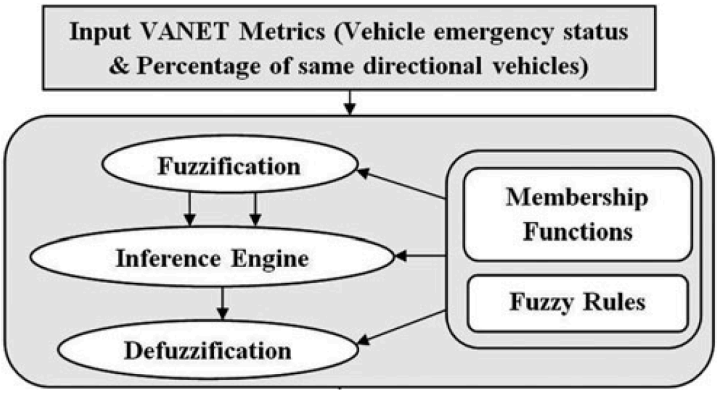


Figure 6.15 Fuzzy logic components

Furthermore, the fuzzy inference system needs: fuzzification, knowledge rules and defuzzification. A group of rules can be used to represent the knowledge base for performing the control action in a semantic form.

The inputs of the fuzzy controller that will adjust the amount of redundancy are:

- SNR of the communication channel.
- Vehicular traffic density.

These two variables allow optimizing and adjusting the amount of redundancy and, consequently, improving the network performance (bandwidth utilization). The overall process involved in calculation of redundant packets is described next.

6.2.6 Fuzzification of Inputs and Outputs

In order to control the amount of redundant packets produced by the NC operations, the source and intermediate nodes encode m received packets out of all K packets it has received so far [203]. The ratio m/K is refereed to as the coding density (output parameter) and it is used to adjust the degree of redundancy as a function of two input parameters (traffic density and SNR).

The redundancy controller is based on a fuzzy logic component. This is because it simplifies the design of a stable controller, which makes use of two input variables. In addition, modifications are easy to perform to the rule inference system in order to enhance the performance of the controller. One just needs to add specific rules. It is worth to mention that the developed controller is implemented in the source node and intermediate nodes in a distributed way. We are using traffic density and communication channel SNR parameters in our redundancy controller (Figure 6.16). The reason for this selection is that higher traffic density variations in vehicular environments yield to high communication channel fluctuation and variability of the SNR. All nodes in a distributed way estimate the channel condition in the form of channel SNR. Furthermore, each node calculates the traffic density based on the proposed method in [204]. Nodes count their neighbour nodes based on the handshaking between them and then adjust the value of redundancy using the fuzzy controller. Moreover, the triangular and trapezoidal functions are chosen as membership function since they have been extensively used in real-time applications due to their simple formulas and computational efficiency.

6.2.6.1 Fuzzy Inference Engine

The fuzzy controller was designed considering nine rules that are presented in Table 6-2. Channel SNR is presented on the horizontal axis and traffic density is presented on the vertical axis. In order to demonstrate the correct behavior of the fuzzy controller, one rule is used to show how the algorithm works and the outputs of each rule are combined for generating the final controller decision [195].

Table 6-2 Rule-based redundancy control

Traffic Density	Low SNR	Medium SNR	High SNR
Sparse	Very High	High	Medium
Dense	High	Medium	Low
Very Dense	Medium	Low	Very Low

Consider a rule "If SNR is Low and Traffic Density is very sparse, the change in coding density is very high" as an example to calculate the output of the system. Our fuzzy inference system considers a case where SNR is 2.22 dB and the traffic density is 12.9 car/km (this value is increased to 13). The output is 0.891(0.9). This value (0.9) represents the coding density, that is, for instance, the case when intermediate nodes encode 9 packets (in our simulation 10 is total number of packets in each generation) out of all received packets. This high redundancy is because of having low values of SNR and sparse traffic density. This output is achieved by using Mamdani's fuzzy inference method [195]. Figure 6.17 depicts the behavior of the redundancy controller.

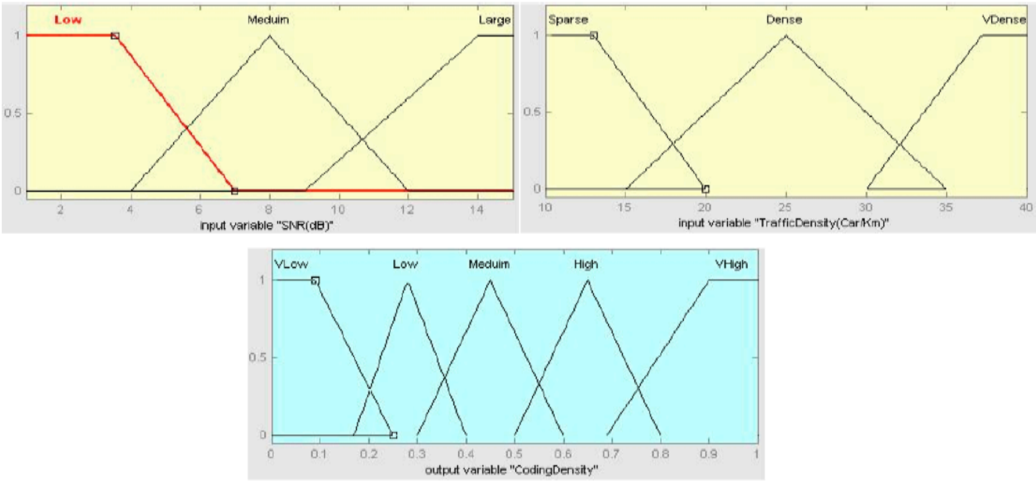


Figure 6.16 Membership function for a) SNR, b) Traffic Density, c) Coding Density

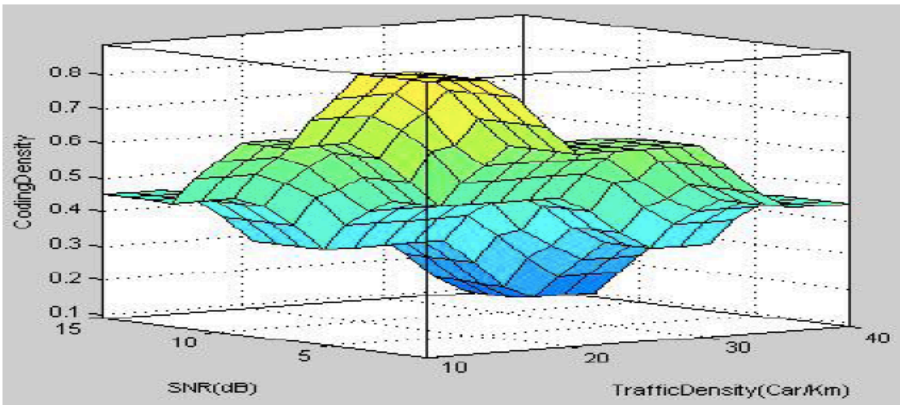


Figure 6.17 Correlation between input and output parameters

6.2.6.2 Defuzzification

Defuzzification refers to the way a value is extracted from a fuzzy set as a representative value. Our fuzzy controller takes the centroid of area strategy for defuzzification. This defuzzification method is based on (eq. 4), as follows:

$$(eq.4)\ R = \frac{\sum_{All\ Rules} \chi_i\, \gamma(\chi_i)}{\sum_{All\ Rules} \gamma(\chi_i)}$$

where R is the redundancy, xi is a specific parameter and Y(xi) is its membership function. This method is the most popular defuzzification strategy.

6.2.6.3 Delay Constraint of video transmission

Consider M as the maximum number of video packets that can be transmitted in a communication session (source to destination) and satisfy the maximum allowed delay T (Table 6-3). The overall video packet delay (T) considers the following three components:

- **T_{NC}** : the time that it takes for encoding and decoding of generations in NC operations.

- **T_{MDC}** : the time needed to split the video file at the source and merge it at the destination.

- **T_t** : the delay introduced due to video packet transmission from sender to receivers, including the sum of packetization delay, propagation delays over intermediate links and queuing delays in intermediate vehicles.

In order to fulfill a video decoding deadline constraint, the worst case scenarios should be considered when calculating the maximum allowed video packet delay.

Thus, it is necessary to calculate each delay during a communication session. The delay caused by NC operations are considered at the encoder and decoder $T_{NC} = T_{Encoding} + T_{Decoding}$. This encoding and decoding delay depends on the number of native packets in each generation. The descriptions are divided to a number of generations of the same size, referred to as the native generations (g_1, g_2, g_3,.., g_r). The time needed to encode the native packets of each generation is denoted by $T_{Encoding}$ while $T_{Decoding}$ is the time to decode all collected coded packets at the receiver. The decoding delay is shown in (eq. 5 and eq. 6) under the assumption that the packets are uniformly and periodically received.

$$(eq. 5)\ T_{Decoding} = T_{Wait\_combinations} + T_{Retrieval}$$

$$(eq. 6)\ T_{Decoding} = \left(\frac{m}{n} \cdot \frac{1}{f \cdot n}\right) + z$$

For encoding packets at the source and intermediate nodes, it is important to collect enough number of packets belonging to the same generation to produce coded packets. Therefore, the encoding time $T_{Encoding}$ depends on the number of packets of a given generation. Table 6-3 shows all parameters related to delay. The main parts of the MDC source coding are the splitter and merger which provide particular processing of the video sequences at the source and destinations respectively. Thus, the time needed for MDC operations is as follow (eq. 7):

$$(eq. 7)\ T_{MDC} = T_{Splitter} + T_{Merger}$$

Therefore, delay parameters should be chosen to satisfy the following delay constraint (eq. 8):

$$(eq. 8) T \geq T_{MDC} + T_{NC} + T_t$$

Table 6-3 Delay parameters

Parameter	Description
n	Total number of packets in a given generation
m	Encoded packets out of n native packets
f	Frame rate (number of frames per second)
$1/(f \cdot n)$	Inter-arrival time between successive native packets
m/n	Coding density
z	Time needed to invert the coefficients matrix
Twait_combinations	Time needed to collect enough coded packets (nonlinear combinations)

6.2.7 Performance Evaluation

6.2.7.1 Simulation Setup

In this section, the simulation setup to evaluate the performance of the proposed reliable joint coding approach is introduced. The simulation is performed using NS-2 simulator (with default values for MAC and physical layer configuration parameters) integrated with Evalvid [205]. Furthermore, a MDC simulation has been carried out to generate video descriptions [206]. Two different video streaming approaches have been tested:

- Joint Source-Network Coding approach (JSNC)
- NC approach

The fuzzy logic algorithm was implemented using the jFuzzyLogic open source software [259]. Essentially, the algorithm implemented here (published in [GP9]) is the same as used in [GP5]. The main differences are in the definition of the membership functions, required parameters and specific algorithm integration for each scenario (it is a piece inside an heuristic for adapting the data redundancy or becon frequency). Actually, the architecture for implementing this decision module is the same.

The SUMO tool [207] has been employed to create the vehicular scenario, and to generate mobility traces of the simulated vehicles. SUMO interposes vehicles in each lanes at a given traffic rate. This mobility model allows us to simulate vehicular scenarios such as speed deceleration/accelerations and different vehicular traffic conditions. The vehicle that acts as video source is generated first in one route. In our simulations, each route has its own parameters such as speed and acceleration.

All vehicles in our simulations have a transmission range of 200 meter and a capacity of 6 Mbps per channel (symmetric). The roadway used is a two-lane road of 5 Km length. Vehicles access the road according to a Poisson distribution and travel at a speed between (10-23) m/second. The simulation ran for 100 seconds, resulting in a total of 125 vehicles generated. In the simulation, a CIF (352x288) MPEG2 video sequence (Foreman) was used with. The video sequence was 300 frames long. The encoder includes three types of frames: Intra frames (I-frames) that are independent on other frames, and inter frames that use either P-frames or B-frames temporal prediction from other frames. A number of mutually dependent frames form a group of pictures (GOP). In our experiment, the GOP size was set to 12 frames.

6.2.7.2 Simulation Results

We considered three different metrics in our evaluation:

- Average Peak Signal to Noise Ratio (PSNR).
- Protection efficiency: average packet loss rate, which represents the robustness of the protocol against packet losses.
- Application layer throughput: this throughput depends upon packet losses at the lower layers.

Figure 6.18 a) shows that the overall video quality is higher for the proposed approach than the obtained from the NC approach results. When vehicles are travelling at low speed, the value of PSNR is slight different for both approaches. This is because at low speed the wireless channel is stable and intermittent connectivity occurs less frequently. As the speed of vehicles increases, the disconnection rate between the source vehicle and the receiver increases more frequently as well as the high channel variations. Obviously, this situation leads to packet loss. A coded packet loss in one generation causes multiple packet loss as well in subsequent generations due to the video coding approach (GOP). As result, the average video quality of the proposed approach is higher than that of the NC approach.

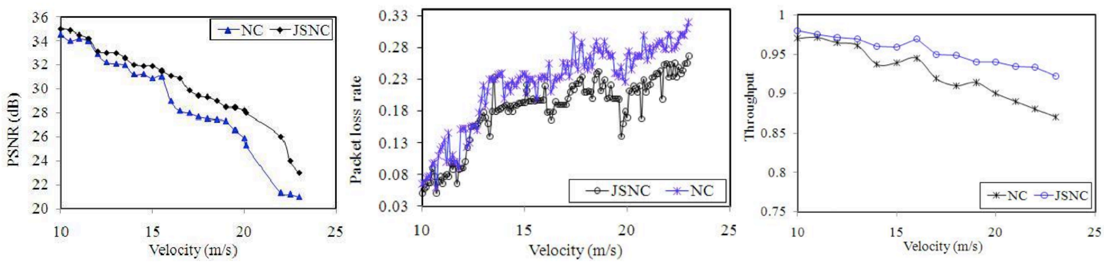


Figure 6.18 a) Video quality for different vehicle speeds, b) Instantaneous packet loss rate for different vehicle speeds, c) Application layer throughput for different vehicle speeds

To compare the packet loss rate of the proposed approach with the NC approach, the video source transmits video packets at a given period time. The source vehicle transmits the video coded packets from source to the receiver applying NC operations. In the next simulation we repeated the video transmission by applying the proposed approach. After that, during offline analysis, we calculated the average packet loss rate through a comparison of the receiver and source trace files. We observed that the instantaneous packet loss rate is increasing in both approaches and the overall packet loss rate is lower for the proposed approach than that for the NC approach as depicted in Figure 6.18 b).

Initially, there is small difference of instantaneous packet loss values for the proposed approach and the NC approach due to limited channel variations (less bit error rate and fading of the channel) as well as less frequent link disruption. Consequently, the number of lost packets decrease. As mobility increases, the channel instability (high bit error) and intermittent connectivity increase. As result, packet loss increases in both approaches. However, the NC has more coded packet loss than the proposed approach at high speed. This is because one lost packet will probably generate multiple losses in subsequent packets. However, MDC increases error resiliency of NC by generating multiple descriptions. We observed a similar trend when measuring application layer throughput as we vary the speed from low to high in Figure 6.18 c). The NC approach shows a decrease in throughput due to high speed of vehicles. This leads to high coded packet loss. Regarding application layer throughput, it almost does not change at high speeds. This is because high coded packet loss (due to high speed) at the lower layers reduces effective application layer throughput as well as increases bandwidth variations.

As seen in this section, the NC operations are implemented above the MAC layer while MDC operations are embedded at the application layer. The MDC descriptions are adapted with NC generations. This two stage coding mechanism increases the robustness of video streaming in vehicular networks. Moreover, the NC redundancy is adjusted using a fuzzy inference system controller. Then, the tuning of the amount of redundancy was based on vehicular traffic density and SNR of the communication channel. NS-2 extensive simulations were successfully carried and showed significant gains in terms of average video quality, application layer throughput and average packet loss rate. The video streaming with NC has been compared to the proposed joint coding approach and the proposed scheme gives better performance.

6.3 Reliable Video Streaming over VANET using FEC mechanisms

FEC mechanisms can be applied to different symbols with in a packet (bit, byte, a block of bytes). In addition to that, FEC can be used in different Open System Inter-connected (OSI) layers such as application, network, Physical and MAC layer. In this thesis, FEC is applied to complete packets in application layer, which is known as packet level FEC [242]. This technique is able to recover packet losses without retransmission request (retransmission of lost packets in large scale video transmission is often not practical) [241].

Packet level FEC is a mechanism for protecting RTP payloads against packet errors by adding specific FEC redundant data to the transport stream. Furthermore, the packet level FEC is combined with the interleaving technique to increase the error protection efficiency. This is because the combined schemes can scramble correlated burst packet losses and recover them. However, interleaving requires additional delay when interleaving depth (m) and block size (n) is too large. Thus, it is necessary to adaptively monitoring the packet loss pattern of wireless channel, in order to apply an effective interleaved FEC protection.

In VANET, the position and distance between vehicles are variable. In this case, the packet error rate and loss pattern also are variable between source and destinations. Actually, approaches like [243][244] use interleaved FEC to protect wireless channel against burst packet loss. However, these approaches are error resilient to static wireless networks, position variation of wireless nodes were not taken into consideration. Still, most of the works focusing on static wireless or low mobility wireless networks. Therefore, an error resilient scheme is needed to be addressed for reliable video geocasting in urban vehicular traffic condition.

Next, we propose a scheme based on packet level interleaving FEC for reliable video geocasting over VANET [GP4]. The proposed error resilient scheme is able to tackle the above mentioned issues. The first phase of this process generally consists of the system architecture of proposed error resilient scheme. The aim of this architecture is to demonstrate the functions of the proposed scheme. The second phase consists of demonstrating real time video streaming and RTP/RTCP protocols. This RTCP report provide accurate feedback that indicates for each transmitted packet per frame, whether it was loss or received. In the third phase, we adaptively vary the amount of redundancy according to the wireless channel condition. This adaptive variation is based on RTCP report of farthest vehicle. The fourth phase is based on the validating of proposed scheme based on simulations (NS 2), showing consistent gain in PSNR as well as maximizing the protection efficiency in urban vehicular scenario.

6.3.1 Video Transmission over Wireless Networks

Recent years, error resilient mechanisms based on Automatic Retransmission Request (ARQ) and FEC are widely used to correct video stream errors [242][245]. The extensive studies in wireless multimedia communications have shown that the effect of FEC scheme on packet errors yields better bandwidth utilization and lower delay than ARQ. Thus, FEC schemes are increasing error resilient in wireless multimedia communications [243]. The most popular FEC scheme is Reed-Solomon (RS) codes to generate packet level FEC blocks. A FEC coder is a block coder that takes a block of **k** source packets as input and produces **n** FEC packets as output $(n > k)$. In the receiver, the original video data can be regenerated, if the number of packet errors is less than decoding threshold for the FEC code. On the other hand, interleaving techniques are used to convert burst losses to equivalent numbers of isolated packet losses. In this way, it can be used to increase FEC efficient recovery.

However, the efficiency of FEC recovery depends upon both the size of FEC blocks and number of source packets that are interleaved and scrambled. Admittedly, it is necessary to estimate the channel dynamicity through the network level metrics (packet loss rate) to change the amount of FEC redundancy and interleaving depth. In [246], authors proposed an adaptive FEC mechanism for minimizing end to end video distortion. In this mechanism, the decision of packet transmission is based on bandwidth rate constraint. In [247], authors studied an error control adaptive mechanism based on the packet traces of Wireless LAN. In [248], authors proposed efficient packet-level interleaving with RS coding. Furthermore, literatures also suggest an algorithm based on delay-aware for minimizing delay generated due to interleaving. They have also shown that packet-level interleaving with FEC results in better video quality.

Some literatures also introduce the RTCP adaptive feedback which is used to optimize the amount of FEC redundancy. However, none of them look into the amount of packet loss variation which is carried by RTCP feedback with different position of wireless nodes. In vehicular networks, the mobile nodes represent vehicles that are travelling in a higher range of speed. Hence, the network topology changes very fast. This mobility causes on one hand, different distances between source and destinations (in the geocast region), on the other hand different speed of destinations. In both cases the packet loss pattern among different receivers are different with respect to the source, thus the feedback RTCP from different receivers have different packet loss. Thus, this distance variation leads to different packet loss pattern, hence different perceived video quality of receivers.

6.3.2 Geocasting over VANET

Inter-vehicle position based communication has been found to be more suitable for VANET environment. In this case, the physical positions of vehicles are required, in order to facilitate communication in high speed vehicular scenarios [251]. Thus, cars are assumed to be provided by Global Positioning System (GPS) - enabled to know their geographic position. In addition to that, each vehicle periodically broadcasts a beacon to obtain the information of neighbour nodes (see our contribution [GP5] and APPENDIX V). This information enables the geocast services over VANET. Geocasting is basically a location based multicasting [249]. The objective of geocasting is to deliver information to a group in a specified geographic area.

The basic access method IEEE 802.11 is based on Distributed Coordination Function (DCF). A node is initiating the transmission after the sensed channel is idle. This mechanism is based on CSMA/CA protocol to schedule the medium accessing. Although DCF is simple and efficient, it cannot support Quality of Service (QoS) for multimedia applications. It is for this reason, 802.11e [250] is developed for better access mechanisms and supports QoS. The IEEE 802.11e Enhanced Distributed Channel Access (EDCA) is designed to enhance the 802.11 DCF by providing the required QoS mechanisms [252]. In EDCA (as in DCF) the multicast traffic is defined as an unreliable service, i.e., it does not include the use of Acknowledgement (ACK) frames.

6.3.3 Proposed architecture

The architecture of the proposed scheme for video geocasting over VANET is shown in Figure 6.19. The basic idea of the proposed scheme architecture lies on implementing streaming of video from single vehicle (Source) to a group of vehicles (Receiver 1 and 2) in the urban area scenarios (see Figure 6.19). The source generates a video file that is stored in raw video format. The encoder receives the raw video file, and then it encodes to bit streams. The frames of coded bit streams become input to FEC coder, which applies coding redundancy at a specified rate. This amount of redundancy (n,k) depends on RTCP feedback. Next, as an error resilient scheme, interleaving is applied on a group of packets. After interleaving technique, RTP packets are sent over vehicular network wireless channel. The proposed scheme is implemented in urban area of vehicular scenario, in which vehicles are running with moderate speed.

RTP media streams are transmitted through a wireless channel. The channel error causes packet corruption in Receiver 1 and 2. However, the number of packet losses is different due to different distances of vehicles with respect to the video source. Afterwards, both receivers depacketize the RTP video packets. After that, the video packets are deinterleaved to scramble the burst losses, to separate losses. Next, FEC redundant data is used to correct corrupted packets in a given block of packets in a specified frame. Next, it can be compared with the source video data. In this architecture, the video source vehicle receives the RTCP feedback from each receiver to accurately capture the packet loss.

RTP is designed to be independent of the underlying transport and network layers. In order to enable real time transmission and play out at the receiver, the RTP packet header carries information like sequence numbers and time stamps. A receiver uses the sequence number to detect loss packets and time stamp to determine when to play out received data. On the other hand, Real Time Control Protocol is used to monitor the QoS of the ongoing session. This feedback signal carries statistics information (packet loss, round trip time, and jitter) related to the receivers in the wireless session.

The intended receivers flood RTCP messages to the network. However, we used the proposed method in [253], which is useful for single source and a group of receivers. Thus, the unicast RTCP feedback from different nodes are received by the video source rather than it is flooded into the whole network.

The periodic RTCP report is the critical parameter that must be considered in the FEC adaptation system. The reason for this is that channels between source and receivers exhibit varying packet loss rates over time. The frequency of the receiver reports, which give to the sender an estimate about the network loss rate and other parameters, may reduce the responsiveness of the FEC scheme, leading to suboptimal FEC efficiency. A high frequency would increase the responsiveness with channel variations. This leads to high variations between successive measurements and instability due to excessive feedback traffic overhead. Low frequency would have good stability and low overhead but poor responsiveness. In this work, the receiver sends back the feedback report after a periodic amount of time to achieve the trade-off between source responsiveness and network overhead.

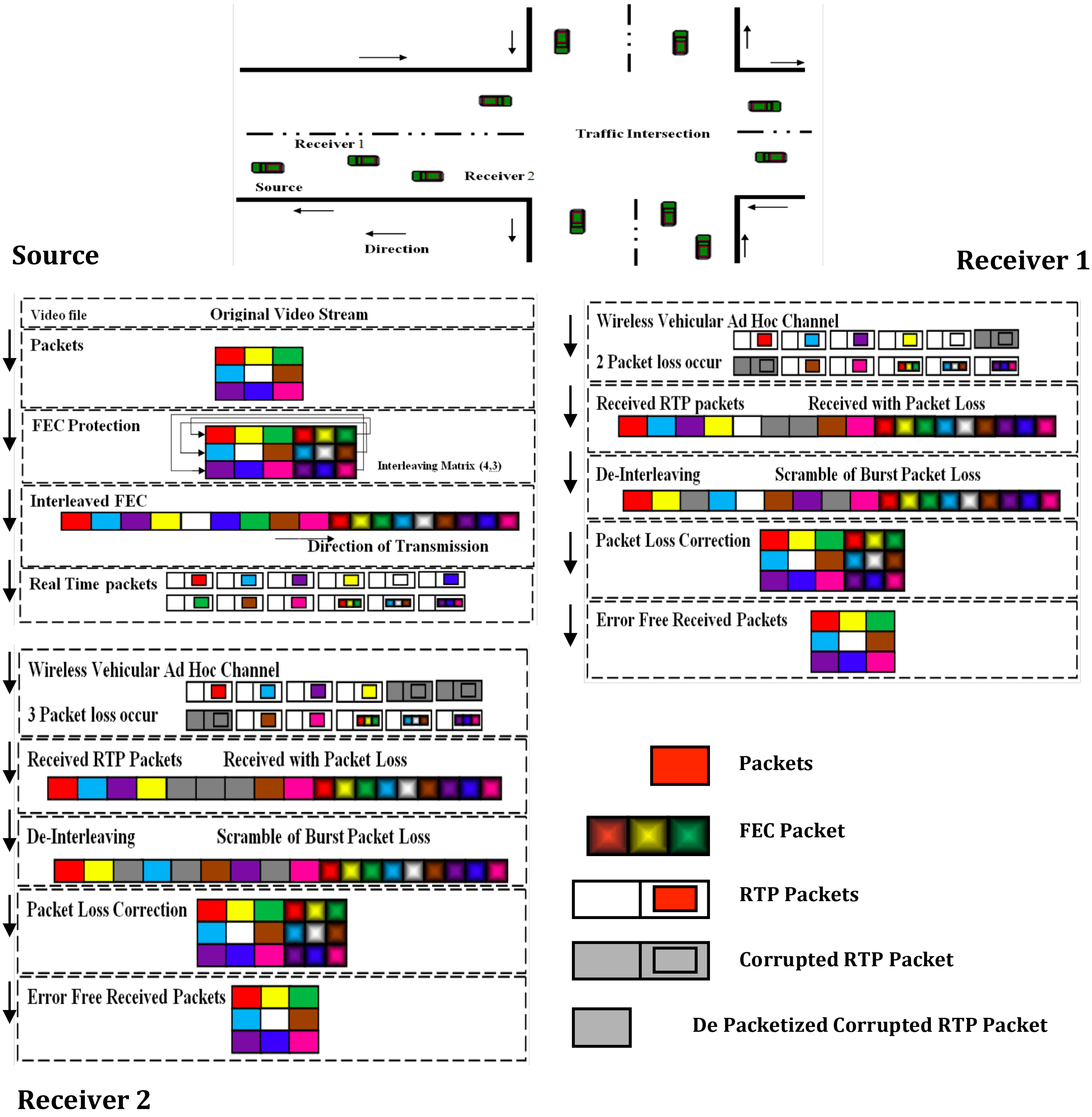


Figure 6.19 Video streaming architecture for VANET

In vehicle-to-vehicle video communications, the video source receives periodically all the RTCP reports of the receivers. The packet loss rate, which is carried by theRTCP report, depends on the distance between source and receivers. Therefore, the proposed scheme uses the receiver reports based on vehicular position with respect to the source. This report is used to update the amount of redundancy and interleaving level to derive effective QoS parameters. Packet level FEC with interleaving can provide reliability for video geocasting over error prone vehicular wireless channel. In the next section the proposed error resilient scheme is discussed.

6.3.4 Vehicles and Communication Model

Vehicles are assumed to be equipped with a Global Positioning System (GPS). Vehicles also transmit periodic beacons [GP5]. Therefore, vehicles know their own geographic location and can be easily synchronized.

Each vehicle broadcasts a beacon (MAC broadcast: which are used for local topology sensing only, it is not re-broadcasted) with a range of about 200 meters. The content of these beacons are helpful to allow vehicles to communicate and synchronize. This beacon is used in our error resilient scheme to establish communication between video source and receivers in geocast region.

The beacon is broadcasted twice a second and contains vehicles information such as: address (ID), location and speed. Each vehicle maintains a data base to store information of vehicles within its transmission range (as is introduced also in [GP5], they can be used for exchanging context information). Upon receiving a beacon, each vehicle updates its data base with new vehicles information. The database is checked every second by the vehicle in our simulated scenario.

6.3.5 The FEC Estimation Algorithm

The summary of the proposed algorithm is shown Figure 6.20. All vehicles in the same communication range know their position and speed, the beacon can be used to get these informations. In our simulation scenario, the source and receivers are always in the same radio range. In other words, there is no intermittent connectivity between source and destinations. When source receives RTCP feedback from all receivers, first it checks the distance of receivers (we have experimented this technique, but it creates a large amount of overhead, thus, we have compared the distance of receivers based on feedback packet loss rate). Then, it selects the RTCP report of the farthest node (highest loss). Next, it extracts the packet loss rate (line 1,2). If packet loss rate is bigger than a threshold, FEC redundancy is not applied to video packets. The reason of this are: (1) low value of Peak Signal to Noise Ratio (PSNR) to guarantee acceptable video quality, (2) to improve utilization bandwidth (line 3,4,5,6).

The efficiency of FEC recovery is reduced, when the number of packet loss is greater than the calculated FEC redundancy (which is relying on the previous RTCP feedback report). Therefore, the new value of feedback packet loss report is compared with the calculated FEC redundancy. If the calculated FEC redundancy is greater than the feedback packet loss report, the value of FEC redundancy remain the same, otherwise the new value of FEC redundancy is calculated and transmitted along with media packet streams (line 7,8,9,10). In this way, we can keep FEC redundancy to protect the original video source data.

After an amount of FEC redundancy is added to real time media packets, the simple interleaving technique is applied as well. This technique is able to dynamically adapt the interleaving delay according to the wireless channel condition. The details on the FEC algorithms can be seen in [GP5].

Algorithm Feedback based FEC estimation

Require: Vehicle x knows their position, speed using beacon
Require: This algorithm is triggered, when source vehicle receives the RTCP report from a group of receivers
Require: Source & destinations are in the same radio range
1: Source receives the RTCP statistics from receivers
2: Source selects RTCP feedback with highest packet loss rate
3: **if** (packet loss rate ≥ threshold)
4: FEC redundant packets is dropped to zero
5: go to **exit**
6: **end if**
7: Source calculates the amount of redundancy packets (h) based on returned packet loss rate (p)
8: **if** (calculated FEC redundancy > feedback packet loss rate)
9: Redundancy = calculated FEC redundancy;
else
Redundancy= feedback packet loss rate;
end if
10: Apply interleaving technique based on equ. 2 and 3
11: **exit**

Figure 6.20 Feedback-based FEC estimation algorithm

6.3.6 Performance Evaluation

In the following sections, through simulation based study, we emphasize the ability of the proposed scheme in terms of higher packet loss resiliency and better perceived video quality.

6.3.6.1 Simulation Setup

In this section, we present simulation setup to evaluate the performance of the proposed error resilient scheme. The simulation is performed using network simulator (NS-2) [255] with added support of EDCA developed in [254] and integrated with Evalvid [256]. Two different FEC schemes have been experimented: one with fixed FEC (proactive FEC: in which the amount of FEC is not changed in the ongoing session) and the other with the proposed protocol (adaptively changing the amount of FEC and interleaving depth). We also experimented video streaming without FEC. We measured PSNR quality of the video after decoding and Packet Error Rate (PER) for different receiver position.

The SUMO tool [258] has been employed to create the urban scenario, and to generate mobility traces of the simulated vehicles. SUMO interposes vehicles in each lanes at a given traffic rate. This mobility model allows us to simulate vehicular scenarios such as speed deceleration/accelerations and different vehicular traffic conditions. The vehicle which is assigned as video source is generated first in one route (each route has its own parameters such as speed, acceleration, deceleration). Afterwards, other vehicles are generated in the other route. This is helpful to control the distance between source and receivers. A vehicular scenario has been experimented, in order to show the effect of different distances on the video quality and packet loss rate.

Table 6-4 summarizes our simulation settings. All vehicles in our simulations have a transmission range of 200 meter. The roadway used is a two-lane road of 3 km length. Vehicles enter the road according to a Poisson distribution and travel at a speed between (40-60) km/hour. The simulation is run for 20 seconds, resulting in a total of 25 vehicles generated.

Table 6-4 Simulation settings

Parameter	Settings
Transmission Range	200m
Run time	20s
Vehicle density	25car/km
Number of lanes	2
Road length	3Km
Communication data rate	6Mbps

In the simulation, the Common Intermediate Format (CIF) [257] Foreman bit stream is used with resolution 352x288 pixels and 300 frames. The CIF Foreman sequence is encoded with a quality 35.68 dB. The encoder includes three types of frames: Intra frames (I-frames) that are independent on other frames, and inter frames that use either P-frames or B-frames temporal prediction from other frames. A number of mutually dependent frames form a group of pictures (GOP). In our experiment, the GOP size is set to 12 frames. Furthermore, the frame sizes of I, P and B frames are set to 25, 50 and 225 frames respectively. Every 0.5 seconds, the sender receives an RTCP packet from receivers located in its radio range with a report of the current Packet Loss Rate (PLR). After that, the source changes amount of FEC redundancy. Furthermore, a 1.5 second play-out buffer is used to reduce the jitter of video frames at the receiver. Thus, if packet arrives after decoding deadline is discarded. In order to make the results more precise, each simulation experiments are executed more than one time with each simulation execution lasting 20 second. Later, all the results are averaged.

In our simulations, the RS (16, 12) FEC code has been experimented to protect the transmitted video file. This FEC code is applied to both fixed FEC scheme and our packet level adaptive FEC schemes. Both error resilient schemes with no FEC are implemented in the same vehicular scenario. During the simulation, one of the vehicles is the video source and the others in its communication range are receivers. We restricted the traffic density to less number of receivers to demonstrate the effect of distance on the packet loss and video quality.

6.3.6.2 Results

In this simulation-based study, the source first applies our error resilient scheme, Adaptive Interleaved FEC-High Packet Loss (AIFEC-HPL) in which the source takes into consideration the RTCP feedback with highest packet loss rate. Next, we also apply Fixed FEC (FFEC) (the FEC protects the media packets without any feedback). Then, NFEC scheme (No FEC: video geocasting without FEC) is experimented in our simulation environment to evaluate the quality of transmitted video. Furthermore, source transmits video RTP packets over real vehicular networks; each packet contains a sequence number that is used for synchronization of media stream packets. If the packet is successfully received by the destination, the destination will write the sequence number into the trace file. Subsequently, the receivers trace files is used to reconstruct the transmitted video file. We can discriminate between receivers distance to the source based on the packet losses in the trace files.

As a consequence of variable speed of vehicles, the distance between source and receivers are varied with time for both receivers. However, the average distance among source to receiver1 is lower than to receiver 2. Therefore, the average video quality is better in Figure 6.21 than Figure 6.22. These figures depict the PSNR values of the video frames for two receivers in the communication range of the source. We can observe that: (1) the video qualities are improved for both protection schemes (AIFEC-HPL and FFEC) against the scheme without any protection. (2) As the distance between source and receivers (vehicles) increases the perceived video quality decreases, but AIPFEC-HPL's PSNR is still marginally higher than that of others schemes. This is because the proposed error resilient scheme: (1) adaptively changes the amount of redundant packets in response of the RTCP feedback of a given receiver; (2) takes into account the RTCP feedback with highest packet loss value. Moreover, we find that AIPFEC-HPL's PSNR is almost similar to FFEC in some cases (at the beginning of video frame transmission). That is because of the short distance between source and receivers at the beginning of geocast session.

Figure 6.23 depicts that the proposed error resilient scheme has a better average PSNR for different distances between source and receivers. The trend shows that the average value of PSNR decreases for three error protection schemes as the distance between source and destination increases. This is because the longer the distance, the higher the packet loss. Hence, this leads to lower perceived video quality. However, our scheme is more reliable to the distance variation effect due to dynamic tuning the amount of FEC and taking into account the farthest node RTCP report.

As the video source received different RTCP messages (feedback), several receiver reports were averaged to get the packet loss rate. In our simulation, this adaptive packet level FEC is called AIPFEC-Aggr. The measured average PSNR values of two adaptive error resiliency schemes are plotted in Figure 6.24, which clearly shows the AIPFEC-HPL scheme has much better PSNR values as the distance increases.

To compare the error recovery capability of the proposed scheme (APFEC-HPL) with other schemes, the video source transmits 5267 video packets in a given period of video geocast session. Firstly, we transmit the video packet from source to receivers without error protection. In the next simulation period, we have repeated the video transmission by applying other error protection schemes. Subsequently, during off line analysis, we have computed the number of loss and recovered packets through a comparison of receiver and source trace files. Then, we have recorded the number of received and lost packets. Our proposed mechanism (AIFEC-HPL) recovered a total of 174 RTP media packets while the FFEC approach recovered 61 and AIFEC-Aggre. recovered 103 packets.

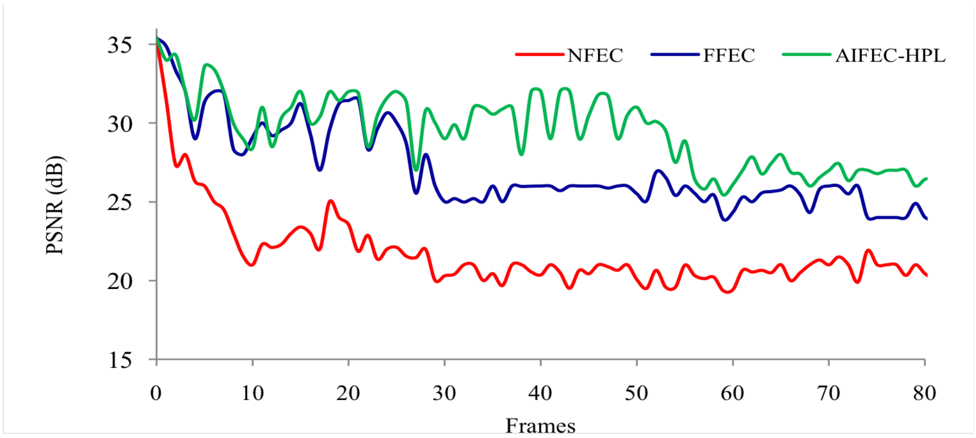


Figure 6.21 FEC Comparison, short distance between source and receiver

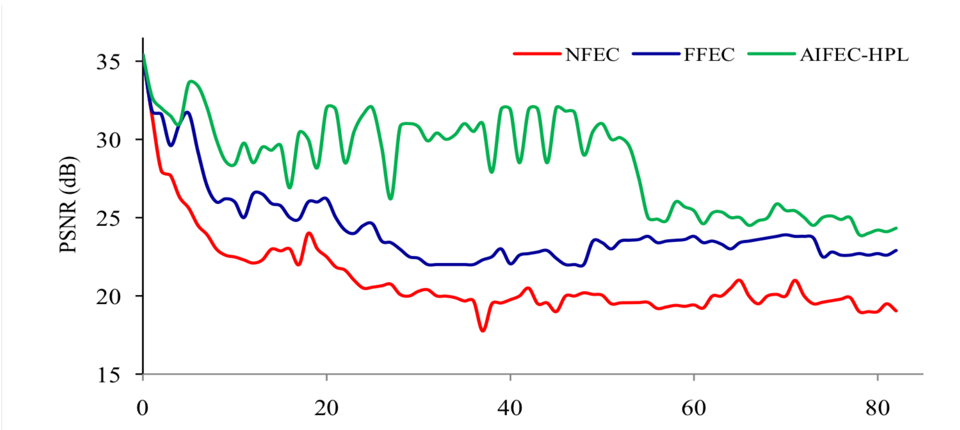


Figure 6.22 FEC Comparison, long distance between source and receiver

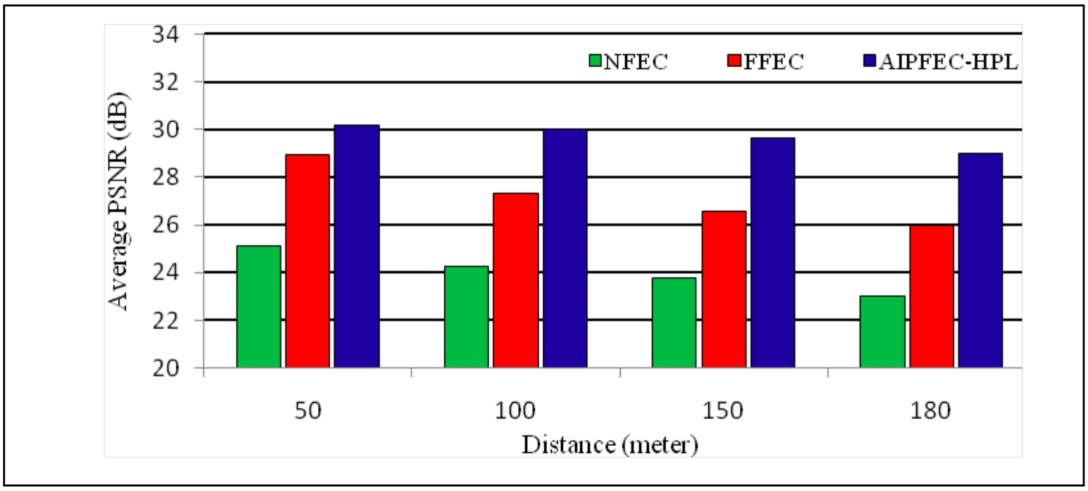


Figure 6.23 Comparison of FEC mechanisms

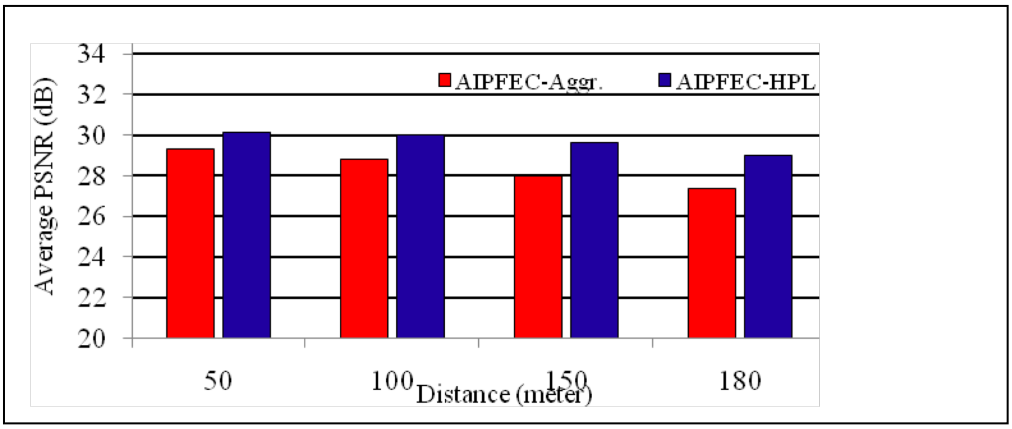


Figure 6.24 Comparison of FEC mechanisms aggregating RTCP messages in the video source

Part III: Service-Oriented and Context-aware Architecture for the Future Media Internet

Chapter 7 Future Internet Flexible Architecture

After seeing different robust, scalable and reliable streaming solutions for different environments applying the studied techniques (FEC, MDC, SVC, NC) in the previous sections, we realized we have proposed different mechanisms at the application level as they are easier to introduce on top of current layered and monolithic Internet. In addition, they are specific implementations that could be reused because when implemented as fully decoupled services. This is a general trend in current Internet, where new services and applications appear running on top of current Internet (over-the-top). However, as said in section Chapter 5, we could represent all these functionalities (as well as all existing ones) as abstract services to enable their automatic discovery and reuse in the Future Internet. To achieve this in a clean manner, we need architecture able to deal with services at all levels, that is, an architecture following Service-Oriented Architecture (SOA) design principles. This section describes the fundamentals of this new architecture that can support all kind of services at all levels. Thus, depicting a more flexible Future Internet.

It must be said that SOA is not a new paradigm. It is very well known and widely deployed in Web Services' environments. However, SOA principles can be applied in a broader context, covering all levels of the communications architecture. As seen in 5.3, some Future Internet proposals seek to create services not only at the application level (like SOA4All [208]), but also at lower levels such as the network. For example, projects like 4WARD define basic functional components, called *netlets* or building blocks, that offer specific operations. Subsets of them will be provided by nodes in the network and should be composed in order to create efficient end-to-end communications according to the requirements of the communication requesters. Herein, this task involves a context-aware service composition process. Figure 7.1 shows a conceptual model of the service composition process. Depending on the type of requester, requirements may vary. For instance, a user can interact with a service provider or two service providers can offer a combined service using each other services under the rules established by an agreement. Typically, the basic requirements of a communication are expressed in terms of QoS parameters. However, requirements can also be specific preferences as well as other desired, or even mandatory, attributes in the communications such as: minimum energy consumption, specific geographical location of a node, minimum price, etc. The more attributes the requester specifies, the more optimised a communication would eventually turn out to be.

Our work proposes a hop-by-hop Constrained-Based Routing (CBR) that establishes end-to-end virtual circuits between the requester and the provider of the demanded end service. CBR is performed during the service discovery process. More specifically it is one of the goals of this work to propose a negotiation protocol that performs service discovery and service allocation as specially designed for service-oriented Future Internet architectures.

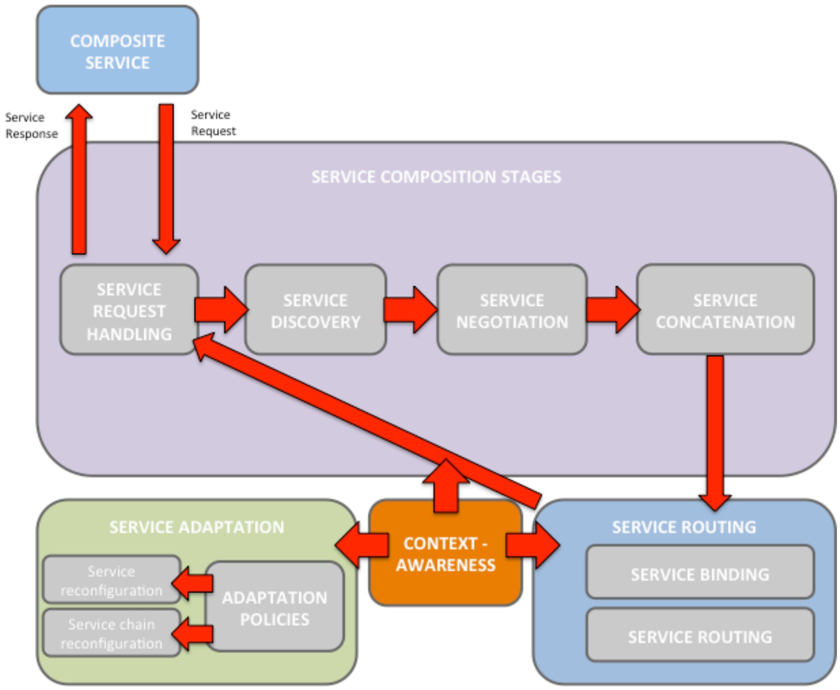


Figure 7.1 Conceptual model of the service composition process

Similarly, along with service information, other context information (network, device, user, service, etc.) is retrieved using this protocol. Here we can adopt the idea introduced in [GP5] where nodes in a VANET exchange context information with their neighbours during the discovery process by means of beacons. This context information would, include characteristics of nodes, networks and users.

Specifically in [GP5], the basic premise of VANET is that in order for a vehicle detect other vehicles in the vicinity. This awareness of other vehicles, can be achieved through beaconing. In the near future, many VANET applications will rely on beaconing to enhance information sharing. Further, the uneven distribution of vehicles, ranging from dense rush hour traffic to sparse late night volumes creates a pressing need for an adaptive beaconing rate control mechanism to enable a compromise between network load and precise awareness between vehicles. To this end, we proposed in [GP5] an intelligent Adaptive Beaconing Rate (ABR) approach based on fuzzy logic (decision algorithm) to control the frequency of beaconing by taking traffic characteristics into consideration. An overview of ABR can be found in APPENDIX V .

Context-awareness is a key feature in QoS provisioning and, consequently, QoE maximization. It allows adapting communications to available network resources and requester needs and preferences. Context knowledge will be a cornerstone enabler of adaptive communications, and it is particularly useful in heterogeneous and dynamic networks. Even more, context data is a fundamental component in this process as it enables automatic systems capable of acting or reacting proactively according to context changes produced during their execution.

Find hereunder an overview of the remarkable features of the proposed architecture. Service discovery, service composition and a costing framework for the Future Internet shall be outlined and discussed next as the main parts of the context-aware service provisioning process and represent the major contributions done in this field in this thesis. Details of these processes are also provided to understand the implementation introduced in section 7.5.

The results of these work were published in [GP6], [GP7], [GP8], [GP13], [GP19], [GP20], [GP21], [GP22] and [GP23].

7.1 Future Internet Architecture Features

Essentially, a new network architecture for the Future Internet should be flexible and adaptable enough to perform as best as possible in all kind of environments, no matter the execution environment, infrastructure support or set of device capabilities. It should provide mechanisms for consuming network services anytime, anywhere, and anyhow whilst being able to integrate all kind of networks and devices.

Thus, we need a shift from strict end-to-end arguments and current stack-based architecture in order to create a new network architecture that provides more intelligence to the network-side whilst still leaving decision-making processes to the end-points. In that sense, SOA principles establish promising foundations for a flexible and scalable architecture for the Future Internet based on basic functional pieces called services. This section aims at providing an overview of the main features of the proposed service-oriented architecture, which allows finding and combining services according to context conditions and, therefore, providing adapted communications [GP13].

- **Oriented to service/resource interconnection and not to machine/interface interconnection.** We propose shifting the focus on network addresses to semantic identifiers of services and resources to facilitate their discovery and usage. Consequently, searches will be semantic and service-oriented**.** We focus on a service-centric approach that allows executing a service search based on user requirements.

- **Integration of QoS into routing and service discovery.** Discovery, establishment and management of routes are based on requested QoS and resource availability. Nodes that receive a service request verify if they can provide the service with the demanded requirements.

- **Network and technology agnostic.** A basic common communication protocol is implemented in all nodes, and used for discovery process and the establishment of virtual circuits. It permits an easy interconnection of heterogeneous networks and devices as well.

- **Semantic identification and addressing of nodes, resources and services**. Service discovery and, hence, routing must be based on the semantic description of the desired service, including security functions. This way, we avoid making explicit addressing (and naming) mandatory.

Besides, existing addresses (locators and identifiers) are treated as another characteristic of the service/node/resource. Furthermore, when used, addressing schemes should be designed to be dependent on the location of entities in a network, but route independent. This semantic context-aware service-derived route discovery approach provides intrinsic support for mobility, multihoming and nomadism.

- **Flexible design and execution**. Network functions must be allocated according to each situation and not in a monolithic way. Thus, functions must be allocated all along the route, executing just the desired functions at each hop, section of hops and end-to-end and applying them just to the desired transmission unit (bit, frame, packet, etc.). Flexible support for different semantic schemes or vocabularies for identifying services, resources and nodes and to describe their capabilities must be also devised. This flexible support must be also extended to different schemes for specifying desired/requested and provided QoS.

- **Context-awareness and dynamic adaptability during execution time.** In order to satisfy QoS and QoE requirements for requested services, context conditions must be thoroughly considered not only during the communication establishment, but also all throughout its duration. The architecture considers the capabilities of the nodes, services and network links to establish new routes and to manage existing ones. As a result, mechanisms to interchange and distribute context information between entities (in ad hoc and structured environments) are to be provided.

- **Native cross-layering.** Cross-layering is inherently provided in the sense of a new disposition of current protocol functionalities, allowing combinations as required and not as a rigid and layered structure usually imposes. Eliminating layers automatically translates into the exclusion of redundant functionalities among them or even counteracting those that could degrade the quality of a communication in a network.

- **User empowerment in service choice and routing**. Given the diversification and popularity of the Internet and the maliciousness of some of their players, users should be provided with mechanisms that give them more control over their communications. This control is reflected in flexible routing and service selection. Thus, a service requester will be able to choose from the set of discovered services the specific service it wants to consume according to his preferences. In addition, and if so desired, the requester may want to specify preferred and trusted entities (carriers, domains, nodes, providers) and blacklists of distrusted or malicious entities. In this way, end-points have a certain degree of control over which routes their communications will follow. A benefit of this approach is that this is performed during the communication establishment and not after.

- **Security.** Security functions must be fully integrated in its design. In our opinion, security must not be an addendum. Hence, service discovery/consumption takes into account available security features. Other points to assure are: data integrity and confidentiality, plus user privacy and confidentiality. Also, we believe that another important characteristic is traceability, e.g., finding the path to the traffic source for a specific communication. As each connection is established for service consumption, traceability can be achieved by univocally tagging each single traffic flow. Therefore, we make use of a unique tag (Session Identifier) to identify and route each traffic flow.

A fundamental design principle followed in the TARIFA project is that the Future Internet architecture must be designed as a flexible architecture, adaptable to context variations. As discussed in [210], a possibility would be to design this network architecture able to work with small devices (e.g. sensors) in limited networks like (e.g. wireless sensor networks). Then, it should be easier to extrapolate it to work in more complex environments, without capacity restrictions (e.g. like wired computer networks). In this work, authors explain they want to avoid the design of complex protocols for wired and backbone networks. Oppositely, the idea is to design simple protocols with the goal to adapt them to environments with restrictions, approach that poses a lot of implementation, design and performance issues.

7.2 Services Framework

The proposed solution considers three basic components: Atomic Services (AS), Atomic Mechanisms (AM) and Composed Services (CS). ASs are individual functions or roles commonly used in networking protocols (e.g. acknowledgment, sequencing, flow control, etc.). These are well-defined and self-contained functions, used to establish communications to create CSs. AMs are specific implementations for each AS, which provide the desired atomic mechanism functionality. An AS can be implemented by different AMs, as shown in Table 7-2. Finally, CSs are a combination of Atomic Services (AS) that work together to provide a more complex service. CS logics needs to be specified in a Workflow (WF) to describe the composition and execution process of functionalities or ASs that could be offered by different implementations or AMs [209] (Figure 7.2).

The composition of ASs consists in discovering, selecting, combining and allocating those services to be executed along the path from a Requester Node (RN) to the End Service Node (ESN) going through different Intermediate Nodes (IN). In this context, a composition process orchestrated by the RN is proposed with the aim to empower the requester's control over the communication establishment. The RN will therefore be able to decide which discovered services best meet its requirements and preferences by centralizing the process of service selection, composition and allocation. This framework is based in our work done within the TARIFA project.

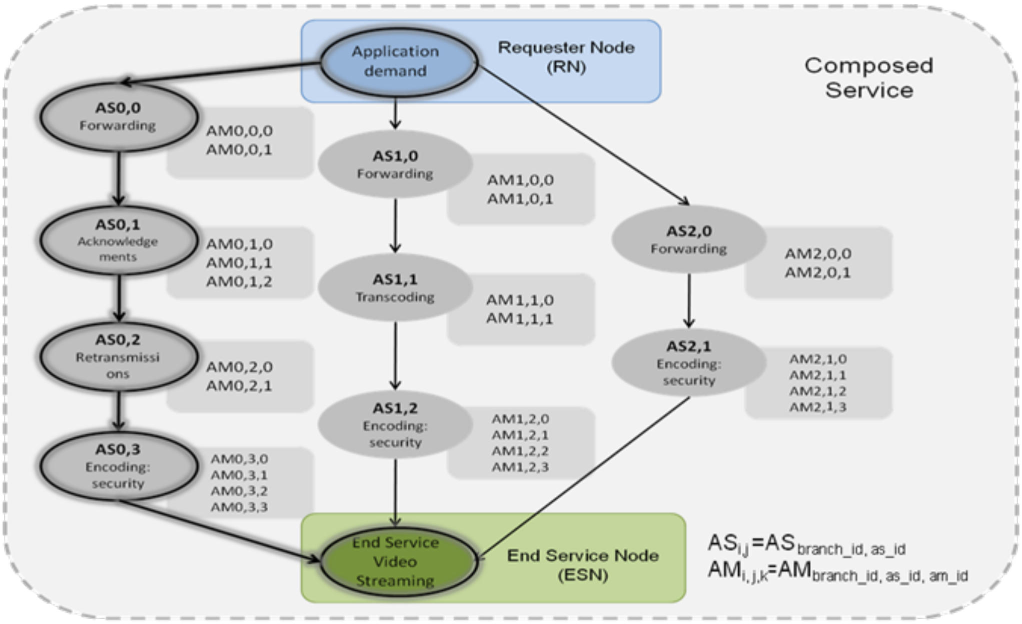


Figure 7.2 Services Framework

7.3 Service Discovery

As described in section 5.5, traditional web service discovery mechanisms are focused on enterprise communications and uses heavy formats based on XML. Pervasive Computing Service Discovery Protocols (SDP) do not provide a unique and integrated solution able to work in a global heterogeneous network. We hereby propose to use an innovative negotiation protocol introduced in [210].

This negotiation protocol allows discovering a service into the network taking into account specific requirements established by the requester. In addition, it is simple enough to be able to work in small and constrained environments such as ad-hoc sensor networks without infrastructure support and larger scenarios with infrastructure support. This negotiation protocol integrates service discovery and service allocation of services, once they are selected and combined.

The service discovery process consists in searching in the network for services under certain conditions, by means of a Communication Request message (*Creq*). In order to specify the criteria that will guide the search, a semantic negotiation protocol is proposed. This message should specify requester's service requirements in terms of:

- Network performance parameters.
- Additional constrains (e.g. geographic, domain restrictions or specific attributes definition for certain services).
- Required functionalities.

Moreover, service requirements may be differentiated in two types of parameters: restrictive and non-restrictive. Restrictive parameters are those that are completely necessary to establish a communication with the desired end service. Non-restrictive ones represent non-mandatory parameters which allow optimizing the communication. Considering the inclusion of service requirements in the request, we propose the following generic definition of a *Creq* message where, QoS_Requirements[i], Context[j], and AS[k] correspond to lists of QoS requirements, context parameters and AS attributes respectively. Additionally, effects can be specified as desired high-level features for the communication, like security or reliability. Resources of a service such as a film provided by a streaming service can be specified as another extra parameter.

Creq *= Session_ID & End_Service_Name & QoS_Requeriments[i](min/max) & Context[j](constraints, preferences) & AS[k](mandatory/optional) [& Effects] [& Resources]*

Using this type of requests, information about the capabilities of nodes is discovered through the network. In this sense, this request can be propagated through the network in different ways, taking into account the network knowledge of the service requester. The default operation performed in a node when receiving a request will be to evaluate if it can provide the service (Figure 7.3). If that is the case, the node answers with a Communication Response message (*Cresp*) that will be transferred through the reverse path. A clear benefit of performing this operation by each node is that it allows to face dynamic and frequent context changes. In the event the node may not be able to provide the ESN, it will propagate the request to its neighbours until the requested service is found.

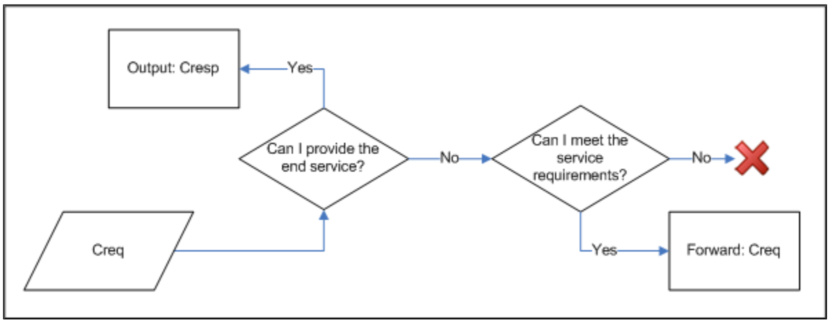


Figure 7.3 Node logics in *Creq* processing

One important issue when propagating a service request is to limit the scope of the request so as not to flood the entire network or propagate it indefinitely. Depending on the kind of network, several approaches can be adopted. In a network without infrastructure support like a sensor ad-hoc network (Figure 7.4 a), a Time To Live (TTL) counter can be set. Oppositely, if the network has infrastructure support (Figure 7.4 b), dedicated directory nodes can perform the discovery process within a network domain. A scalable solution providing interdomain service discovery is a challenging issue, although this topic is out of the scope of this article. Calculating the optimal path whilst considering different constraints is an NP complete problem [211]. However, a near optimal path could be computed taking into account the topology, capacity and context constraints by means of specific heuristics.

It is remarkable that the proposed solution is agnostic to the underlying technology, including networks and devices thanks to the use of well-defined interfaces for service definition. Moreover, the heterogeneity of the network can be faced by consulting nodes capabilities, services and resources thanks to the availability of context information. Each node evaluates the incoming service request and determines if it has enough resources and therefore can meet QoS requirements to deliver the demanded services. Due to the complexity of solving both, service and path selection processes at the same time in a heterogeneous environment, the process can be divided into sub-processes: services discovery and services composition.

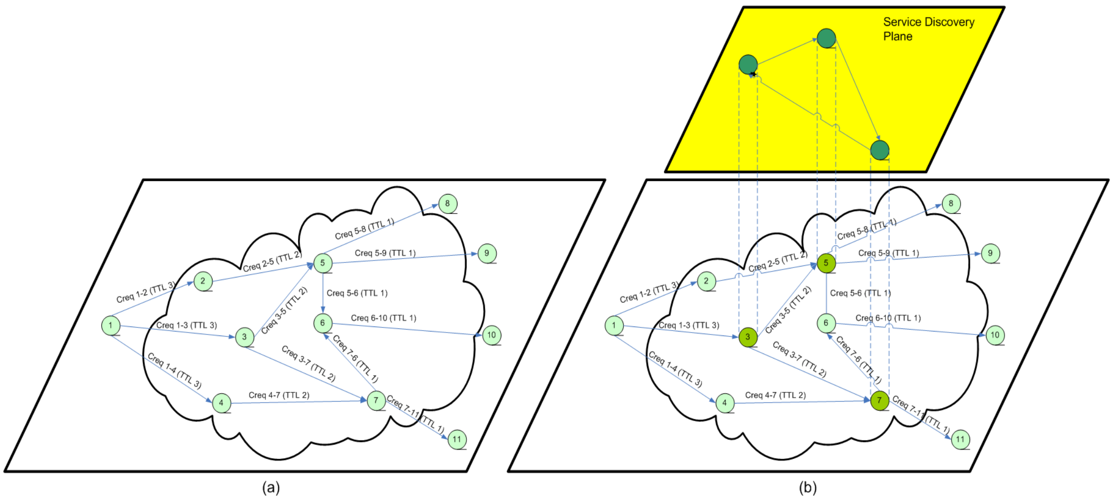


Figure 7.4 a) Network without infrastructure b) Network with infrastructure support

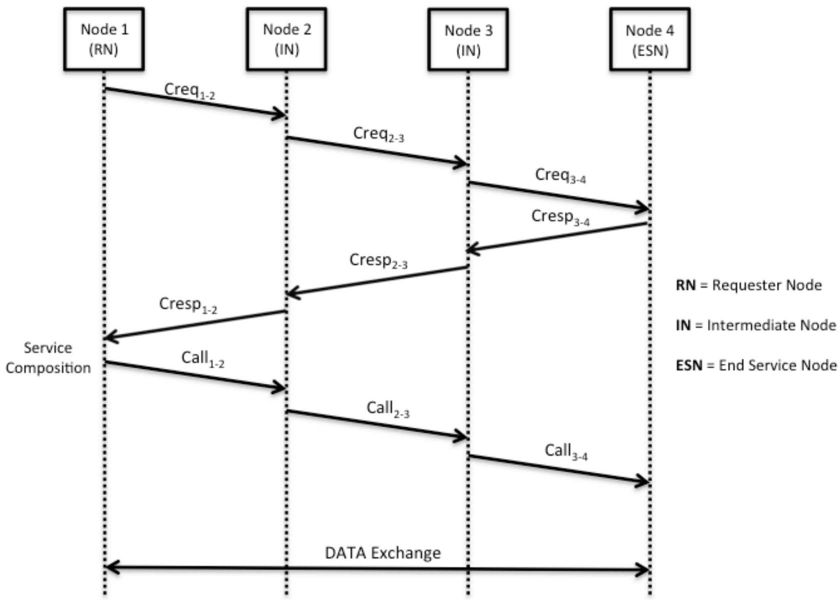
Services discovery is the process of identifying the nodes that can provide the desired end service, as well as the ASs that may be required in the nodes of the communication path ranging from the RN to the ESN. This phase is divided into three steps as well. In the first step, requester requirements are mapped out to a service request. Secondly, the nodes that receive the query evaluate if they are able to provide the demanded service. Finally, context information is consulted to guarantee that the service can be provided under the required QoS parameters.

Service composition process consists in a prioritized selection and combination of the end service and intermediate ASs among all the candidates found during the service discovery. This allows optimizing the network resources during connection setup by selecting those services which minimise the overall costs when providing a service. Service selection must take into account domain policies and the effects that the usage of a service produces over the network operation (e.g. delay, congestion, cost, etc.). This is further detailed in section 7.4.

Furthermore, we propose a generic definition of a Cresp. The requester will receive 'N' response messages that specify 'N' candidate paths. Each of them contains the identifiers of the M path nodes, with their QoS requirements, ASs, AMs. A *Cresp* is defined as:

Cresp = *SessionID, Node[m] (nodeID, QoSRequirements[j], AS[k], AM[l])*

Where, Node[m], QoSRequirements[i], AS[k] and AM[l] are lists of the corresponding parameters.



Protocol Messages/ Node i- Node j	1-2	2-3	3-4
$Creq_{i-j}$	SessionID, EndServiceName, QoS_{e-e}	SessionID, EndServiceName, QoS_{e-e}-QoS_{1-2}	SessionID, EndServiceName, QoS_{e-e}-QoS_{1-2}-QoS_{2-3}
$Cresp_{i-j}$	SessionID, Locator Node4, Identifier Node4, QoS_4,QoS_3,QoS_2	SessionID, Locator Node4, Identifier Node4, QoS_4,QoS_3	SessionID, Locator Node4, Identifier Node4,QoS_4
$Call_{i-j}$	SessionID, $WF_2(AS_2[],AM_2[])$, $WF_3(AS_3[],AM_3[])$, $WF_4(AS_4[],AM_4[])$	SessionID, $WF_3(AS_3[],AM_3[])$, $WF_4(AS_4[],AM_4[])$	SessionID, $WF_4(AS_4[],AM_4[])$

Figure 7.5 Negotiation Process

Finally, the last message considered in the negotiation protocol is the allocation message (*Call*). It is used to specify to each node in the selected path which ASs and AMs must be executed. For this purpose, WF-based representations are used. A communication allocation message (*Call*) can be defined as:

Call = *SessionID, Node[m] (nodeID, WF)*

Where, Node[m] is a list of nodes in the end-to-end selected path. Figure 7.5 shows the whole negotiation process.

7.3.1 Scalability

Scalability is always a critical issue, even more considering current evolution of network technologies, types and number of devices, number and heterogeneity of users connected to the network and amount of information exchanged through the network. Scalability can be limited in network domains because, for instance, the use of flooding mechanisms used during discovery that limit the network domain size and scope. Hence, semantic identification will not scale appropriately outside local domains unless some infrastructure support is available. Besides, clustering strategies are needed to make the semantic routing approach scalable.

Therefore, as proposed in [GP13], attributes related to domains, used as labels, or identifiers to distinguish between administrative domains and networks can be used. The main difference with current Internet scheme is that the information used to define and label different domains is not the same as that used nowadays. Currently, network prefixes and addresses are used. They are tightly coupled with routing protocols. Nevertheless, the FN should works with semantic attributes to achieve such flexibility. FN naming and addressing scheme should not be constrained by their form and meaning as is the case of current IP addresses and domain names.

Hence, any type of attribute is supported and identifiers could take any form. They could be anything that allows identifying and locating a service in a domain. As examples, service identifiers could be a geographical (virtual or not) coordinate bounding the domain area, a random label, an IP address or other legacy/existing addressing schemes requiring no changes in their original architecture. Using one type of attribute or another will depend on the considered scenario and required routing information (e.g. geographical routing will require the use of coordinates).

Thus, high scalability in structured environments could be achieved by means of service discovery that rely on special entities providing the required infrastructure/signaling services whilst avoiding flooding. Some approaches can be adopted such as introducing the use of a semantic resolver instead of performing semantic flooding. The goal of a semantic resolver element would be to process semantic descriptions and map them to identifiers/locators.

And the following infrastructure services/entities would be required according to [210]:

- **Semantic resolver**: resolves semantic descriptions and maps them to identifiers/locators. This includes mapping identifier attributes to locator ones and retrieving information from Context Manager (CM) and resolving it. These resolution services should be designed to concretize semantic identification descriptions into more concise and directly routable identifiers making extensive use of domain related attributes.

- **Context Manager**: manages context information for a specific network domain maintaining a knowledge plane (Tuple space/DHTs/Directory) with identifiers and related profiles. Nodes feed the shared space with its own information on node and service capabilities, whilst monitoring services update link and network state.

- **Dynamic Point of attachment/configuration manager**: provides initial configuration parameters and manages locator assignment (mobility). Provides coupling between client configuration and infrastructure services feeding the CM with the assigned locator and the client with domain-related configuration data.

7.4 Service Composition

The RN to empower requesters' control over the communication will orchestrate services. The requester will always decide which services will be chosen according to the discovered ones. This selection will be done by means of choosing a selection of the ASs that the requester desires to receive considering requirements are met. To achieve this, it is important to have an expressive semantic negotiation protocol to allow matching the requested services with the services available in each node until the end service.

The information discovered in the network is organised in a graph structure where nodes of the graph are the nodes of the network. However, the graph can be represented as a tree of disjoint branches since the *Cresp* obtained from the discovery process can be directly mapped into the structure shown in Figure 5.

This work divides the composition process to create a CS into four phases. To solve the service composition process, we divide the process into the following main sub-processes: Filtering, AM Scoring, AS Composition and Path Selection.

In this research we explored different composition algorithms. Each of them has its own properties in terms of response time, performance, memory usage, accuracy, etc. It remains as future work evaluating and comparing different algorithms for performing service composition and selection tasks. Note that in this thesis we have validated mechanisms to combine different network level services. However, this composition should be very fast in order to not delay the establishment of a requested communication. In this thesis, we implemented in a real prototype (section 7.6) a service composition algorithm based on A* in order to achieve an accurate solution for the different combinations of services we could calculate. Nevertheless, we need to evaluate if we need such kind of flexibility, full flexibility, when dealing with bigger services sets (more variables and cases to evaluate) at network level. Obviously, we can apply this algorithm at higher levels, such as application, where we can benefit from more processing power capabilities. However, at network level we can not assume this. A way of solving this situation is to store pre-established compositions of services in the form of a template and, then, decide which fits better in each situation according to the communication goals. In this sense, the study of the fuzzy logic algorithm implemented, simulated and evaluated in [GP5] and [GP9] (section 6.2) would be considered as a candidate algorithm for deciding which template of services will be used for a communication, which atomic service or atomic mechanism will be used during the composition process or which parameters should be adjusted in order to obtain the desired effect. Using this kind of fuzzy algorithm implies the need to identify the specific parameters, rules and input functions (membership functions and fuzzy rules) to apply according to the problem space to be solved.

7.4.1 Filtering

This phase consists in filtering all received *Cresp* messages according to the requirements specified by the RN. When specifying the *Creq* a constraint or a preference defining a maximum, a minimum or a range of possible costs acceptable by the user can be set up. These filters are represented by specific rules, which can be solved, for example, by means of a Constraints Satisfaction Problem (CSP) method. The filtering process is an inherent part of service discovery. However, a secondary filtering phase is applied once all the communication responses are received in the requester side to validate that all of them meet QoS requirements along the end-to-end path.

7.4.2 AM Scoring

During this phase, the AM that implements each AS is selected according to specific scoring functions. It takes into account different specific attributes related to the AS, for instance the QoS parameters that they can provide and the priorities of the RN. For each AS, a set of possible AMs is scored and the best one is selected. In our preliminary implementation, the AM scoring is an isolated process considering each AM separately regardless AMs interconnection relationships.

We propose to use a generic weighting function (eq. 9) to score the AMs, where weights may vary depending on the preferences introduced by requesters. Herein, it is possible to define trade-offs between different parameters such as the quality provided by the network, requirements and the price to pay for a service. However, scoring functions could be defined for each AS in order to consider specific requirements as shown in section 8.1.3, where a score metric for audiovisual contents is proposed.

$$\text{(eq. 9)}\quad Score_{AM} = A \cdot a + B \cdot b + C \cdot c + \ldots + Weight_n \cdot param_n = \\ = \sum_{i=1}^{n} (Weight_i \cdot param_i)$$

An alternative to the scoring function would be to apply the fuzzy logic algorithm for deciding between specific ASs or AMs. In this case, the system would need to allow the parametrization of each service with specific membership functions. Then, we could run the fuzzy engine like the one introduced in [GP9] and [GP5].

7.4.3 AS Composition

Usually, an operation can be offered by several distinct combinations of ASs. For instance, a reliable service can be provided by means of acknowledgment, error detection and retransmission functions, or by applying forward error correction. Depending on the combinations, provided QoS may vary. Thus, those best suited to satisfy requested priorities will be chosen.

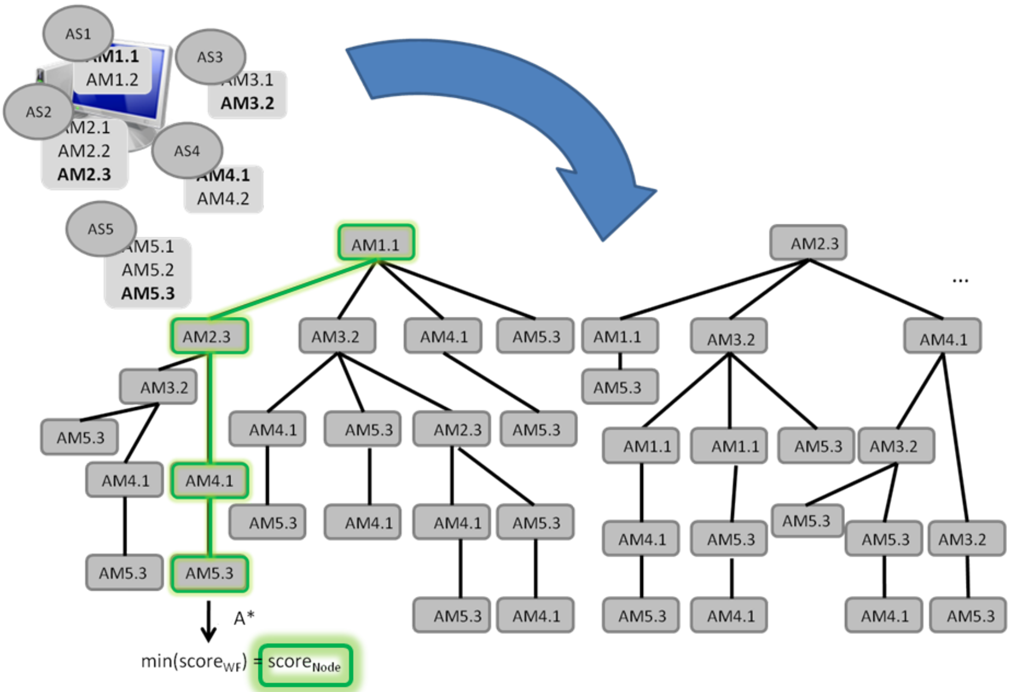


Figure 7.6 AS composition process

Note that the RN composes the services per each node in the path (RN, INs and ESN), and generates the corresponding WFs. Considering the described architecture, once all services are discovered thanks to the discovery process, RN should be able to create a tree graph (Figure 7.6) with all discovered services at each node. Our solution evaluates first the different dependencies at each hop as well as the input and output attributes among ASs, and concatenates those that could be executed within a node in order to satisfy a communication goal. Then, the best branch of services is selected and the final CS is generated for each node. Our first implementation of the service composer component applies A* algorithm [212] for searching the best solution taking into account the nodes with best scoring.

Despite the A*-based algorithm was easy to implement in a real prototype (within the context of the TARIFA project) and offered great flexibility for selecting between many previously calculated combinations of services in different nodes, this algorithm presents some scalability problems and is computational complex. In our tests we used a “reduced” set of services, but in practice, we can expect to have a bigger number of services and, also, mechanisms implementing these services. Thus, other mechanisms must be studied to achieve a feasible tradeoff between performance, response time and output accuracy. As mentioned before, an alternative or complimentary algorithm to optimize the process of service composition could be the adoption of fuzzy algorithms (see [GP5][GP9], section 6.2). This algorithm would allow us deciding between different parameters (aka. services) according to specific goals, parameters and rules. These rules and parameters should be defined by in the system and will determine its final behaviour.

As said, one way to enable fast service composition would be to store pre-established compositions of services in the form of templates. These templates can be scored according to specific parameters to easy their selection in the future and avoid calculating them again. Once a composition is decided (a specific communication is requested), it is very probable that the same or similar communication can be requested again in the future. Thus, it would be useful to store these compositions in order to reuse them and optimize the composition process. Fuzzy logic algorithms could make easier the selection process between different pre-established compositions just by means of determining the set of parameters and rules to apply.

7.4.4 Path Selection

Paths from the RN to the ESN are sequences of nodes containing different ASs, which are capable of reaching the demanded ES. If a cost restriction specified by the RN is at stake, it would be necessary to discard those paths with a higher cost than the demanded one.

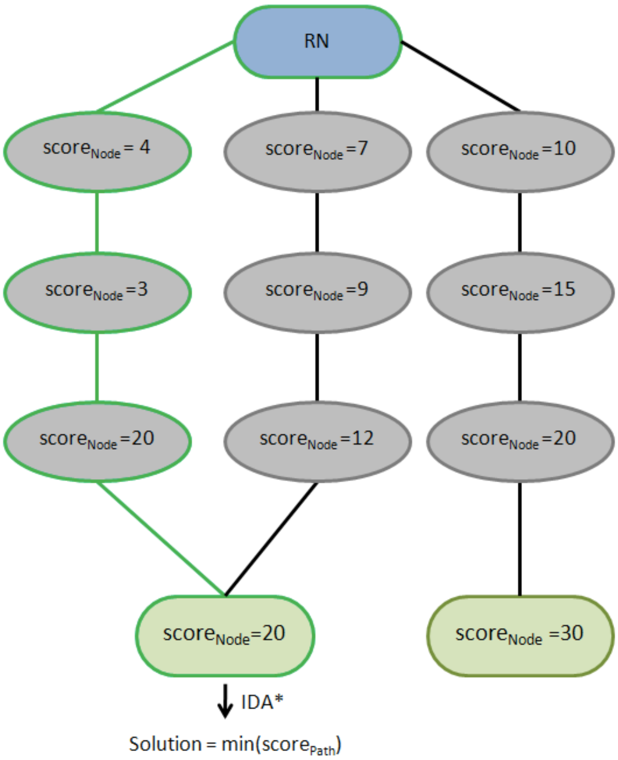


Figure 7.7 Path selection process

When each node is scored, the path is selected using graph theory search strategies to determine the most cost-effective cost path, or branch in this case. Depending on the preferences of the requester, the selected path can offer, for example, the best trade-off between different parameters. For instance, selection could be done according to the lowest delay path, lowest cost per transmitted bit, or one considering a trade-off between both criteria by means of a weighted scoring function (Figure 7.7). In APPENDIX I a description of different composition mechanisms can be found. In addition, APPENDIX IV explains different technical aspects related to the service composition process.

7.5 Enabling Future Internet Service Provisioning

The main goal of the proposed solution is to enable Future Internet service provisioning that permits to meet QoS and QoE requirements. Thus, it will take into account requester needs and context information, including network, device and user features. To show the main features and benefits of this approach we will proceed to describe a challenging use case showing how to provide inherent adapted multimedia communications. Service and content adaptation is an extremely important issue for multimedia communications, especially when it comes to distribution of audiovisual contents in heterogeneous and dynamic networks due to the strong requirements they present in terms of bandwidth, delay, losses, device capabilities, etc. To provide the best QoE to users, QoS needs to be assured, while systems must react against dynamical changes in the network. However, this framework is not only designed to meet QoS and QoE requirements in the provisioning of advanced multimedia services. It also aims to do it in an efficient and transparent manner whilst enriching and improving any existing and future services.

With the proliferation of multimedia capable devices, users demand more and more audiovisual contents. Providing seamless multimedia communications in the Future Internet is and will be a hot and challenging topic for both, industry and academia. Nowadays, Internet users are consistently demanding high quality audiovisual services in a context where operators and service providers are still compelled to deal current technological limitations in order to provide optimised and enriched services, to improve the QoE of users and, in so doing, secure their loyalty whilst making an efficient use of their resources as well as increasing their revenues.

Imagine a user (Ua) who wants to watch a film (F) online from the sofa. Ua accesses the network using a tablet device. At home, Ua uses WLAN 802.11g technology to access to the Internet. Then, it is subscribed to a xDSL Line. User context parameters are described in Table 7-1.

In the network, there are four different streaming services available: service A, service B, service C and service D, denoted by Sa, Sb, Sc and Sd respectively. We assume that these services are placed in different End Service Nodes (ESN), named Na, Nb, Nc and Nd respectively. These are candidate service providers for

Ua as they can offer the service that the user is asking for. Each service in the network is represented by a specific service interface.

Table 7-1 User A (Ua) context parameters (network and device)

User	A (Ua)
Device type	Tablet PC
Screen resolution	1280x768
Supported codecs	MPEG4, MPEG2, WMV
Interface WiFi	802.11g
Internet connection	2Mbps (Uplink)/10Mbps (Downlink)

Table 7-2 summarises the AS and AMs available on these nodes. For the sake of clarity in this use case, we will assume that each node knows which services can be provided. Thus, each node has a local repository with the information shown in this table. In APPENDIX III , a list of network level ASs can be found.

Table 7-2 ASs and corresponding AMs supported by each ESN providing a streaming service

Na		Nb		Nc		Nd	
AS	**AM**	**AS**	**AM**	**AS**	**AM**	**AS**	**AM**
Data_Tx	tx_rx	Data_Tx	tx_rx	Data_Tx	tx_rx	Data_Tx	tx_rx
Data_Rx	tx_rx	Data_Rx	tx_rx	Data_Rx	tx_rx	Data_Rx	tx_rx
Data_Fwd	FIFO	Data_Fwd	FIFO	Data_Fwd	FIFO	Data_Fwd	FIFO
	priority		priority		priority		priority
Seq	incremental	Seq	incremental	Seq	incremental	Seq	incremental
	temporal		temporal		temporal		temporal
ACK	ack	ACK	ack	ACK	ack	ACK	ack
	sack		sack		sack		sack
Retx	retx	Retx	retx	Retx	retx	Retx	retx
Framing	bit_oriented	Framing	bit_oriented	Framing	bit_oriented	Framing	bit_oriented
	byte_oriented		byte_oriented		byte_oriented		byte_oriented
MAC	csma/cd	MAC	csma/cd	MAC	csma/cd	MAC	csma/cd
Error_ Detection &Handling	parity	Error_ Detection &Handling	parity	Error_ Detection &Handling	parity	Error_ Detection &Handling	parity
	CRC		CRC		CRC		CRC
Video Coding	MPEG-1 MPEG-2 MPEG-4 VCx WMV FLV	Video Coding	WMV 3GP MOV FLV	Video Coding	MPEG-4 FLV	Video Coding	MPEG-2 MPEG-4
Audio Coding	AAC MP3	Audio Coding	AAC MP3	Audio Coding	WAV AAC MP3	Audio Coding	WMA MP3

As a first proof of concept, a scenario without infrastructure support is considered. We considered this elementary scenario in order to test it without external support to store information. However, in future work, an approach with a distributed global directory will be undertaken. This approach will require the support of specific nodes, in order to improve service and context information searches, but compatible with the negotiation protocol to assure interconnectivity between heterogeneous networks.

To get the film, Ua sends a *Creq* to its neighbours. This message is propagated hop by hop. Each node evaluates if can provide the End Service that has been asked for or can provide the basic services under the desired conditions, that is, in this case, the QoS parameters that the user requires. At this point, each node applies the logic previously presented in Figure 7.3.

Sa cannot be reached because the INs in path 1 (P1) between the RN and the ESN make the path unsuitable for the communication as they introduce too much delay. Sb, Sc and Sd can be reached through path 2 (P2), path 3 (P3) and path 4 (P4) respectively.

At this point, each node containing Sb, Sc and Sd answers with a Cresp that goes back to the requester through the reverse path.

Once this stage has been accomplished, the requester node evaluates each received Cresp. This is done by applying a service composition algorithm like the one proposed in section 7.4. In this case, Ua can select Sb if he wants a service offered with lowest delay. Sc is the best considering coding compression. However, Sd is the best considering a trade-off between energy consumption (it uses a less demanding codec) and audiovisual quality. Finally, Ua decides to opt for Sd because it meets his visualization preferences and makes a better use of the life of the battery in comparison with the previously tracked down services. As an example, MPEG-2, which is the video codec (AM) available in Sd, requires 8 times less the processing power for encoding and 3 times less the processing power for decoding in comparison with H264/AVC [213] (Figure 7.8a).

Once services are selected for each node and WFs are created, Ua sends a Call through the selected path. This message is the last message defined in the basic negotiation protocol and its main goal is to allocate the services.

The total time to consume a service can be expressed as follows (eq. 10).

$$(eq. 10)\ \ T_{e-e} = T_{composition} + \ (N-1)(3 \cdot T_{propagation} + T_{forwarding} + \ T_{response})$$

Where, T_{e-e} represents the total end-to-end time required from demanding a service to the start in data reception. $T_{composition}$ is the time required for AS selection, composition, path selection and, consequently, per node workflow creation. $T_{propagation}$ is the propagation delay between two consecutive nodes. In the depicted scenario, we assume a constant time for each link. $T_{forwarding}$ is the time required for deciding if a node can provide the demanded end service or not. $T_{response}$ is the time needed to write the Cresp parameters offered by each node. Finally, N represents the total number of nodes in the considered scenario.

As already mentioned, the proposed negotiation protocol allows discovering services whilst evaluating context conditions hop by hop to guarantee the required QoS. To achieve this, the discovery process includes the routing process to the ESN. It is done on a per-hop basis during the establishment of the end-to-end path between the RN and the ESN. Thus, the routing is undertaken considering the context of the network and services to find the best possible path too. Thanks to the service composition process, services can be allocated according to context conditions and specific communications goals.

This use case shows a network with homogeneous INs which perform the same operations. However, the network could be composed of different network nodes with different capabilities and different services. Service composition and allocation allows specifying which services should be placed and executed at each node in order to obtain the best possible communication. Consequently, a node with WiFi interface and wired interface (e.g. copper providing xDSL access) will be able to use different services depending on the features (context) of the interface being used to send data within a node in the selected path. An example would be to use congestion control functionalities in the wired interface whilst avoiding them in the WiFi interface. This is possible thanks to the RBA decomposition of functionalities and SOA-based composition of them according to the specific context where the communication is established, allowing to differentiate between segments of the network and placing services when and where needed.

7.6 Testbed

This section describes the proof of concept implementation of the proposed solution within the TARIFA project and some preliminary results obtained when performing a context-aware service search into a network.

Regarding the generated code, it was migrated from the original modules specified in [210], developed in a System-on-Chip (SoC) CC2430 [214] from Texas Instruments platform. Moreover, code was adapted to run in a Linux-based desktop computer to test the proposed solution in a larger and more complex network. Finally, we extended it with new modules. Mainly, the extensions of the code are: (1) Service composition & allocation module, (2) Search service engine module and (3) Constraint-based routing module.

Table 7-3 Detail of the size of the generated code

Name	Code size (bytes)	RAM (bytes)
Profile module	80422	8656
Network module	246437	4671
Service module	559534	25630
MAC layer	25133	273
OS abstraction layer	212325	76950
Vocabulary API	28332	
Service socket API	42021	
Service composition & allocation	217341	61896
Search service engine	58838	18913
Constraint-based routing	75681	32697
Total	**1546064**	**229686**

The whole development requires a total of memory space (Table 7-3). Regarding hardware and software configurations, they are specified in Table 7-4.

Table 7-4 Specification of the testbed nodes

CPU	Intel Pentium 4 540
Processor speed	3,20 GHz
Number of processors	2
Total number of cores	1 per processor
L2 Cache	1024 Kb per processor
RAM Memory	512 MB
FSB Speed	800MHz
Operating System	Ubuntu 11.04 32 bits

In this testbed (Figure 7.8b) a total number of 13 nodes are used (1 RN, 4 ESN, 8 IN). All nodes are directly connected using several network interfaces configured in Full-Duplex 100Mb/s Ethernet mode.

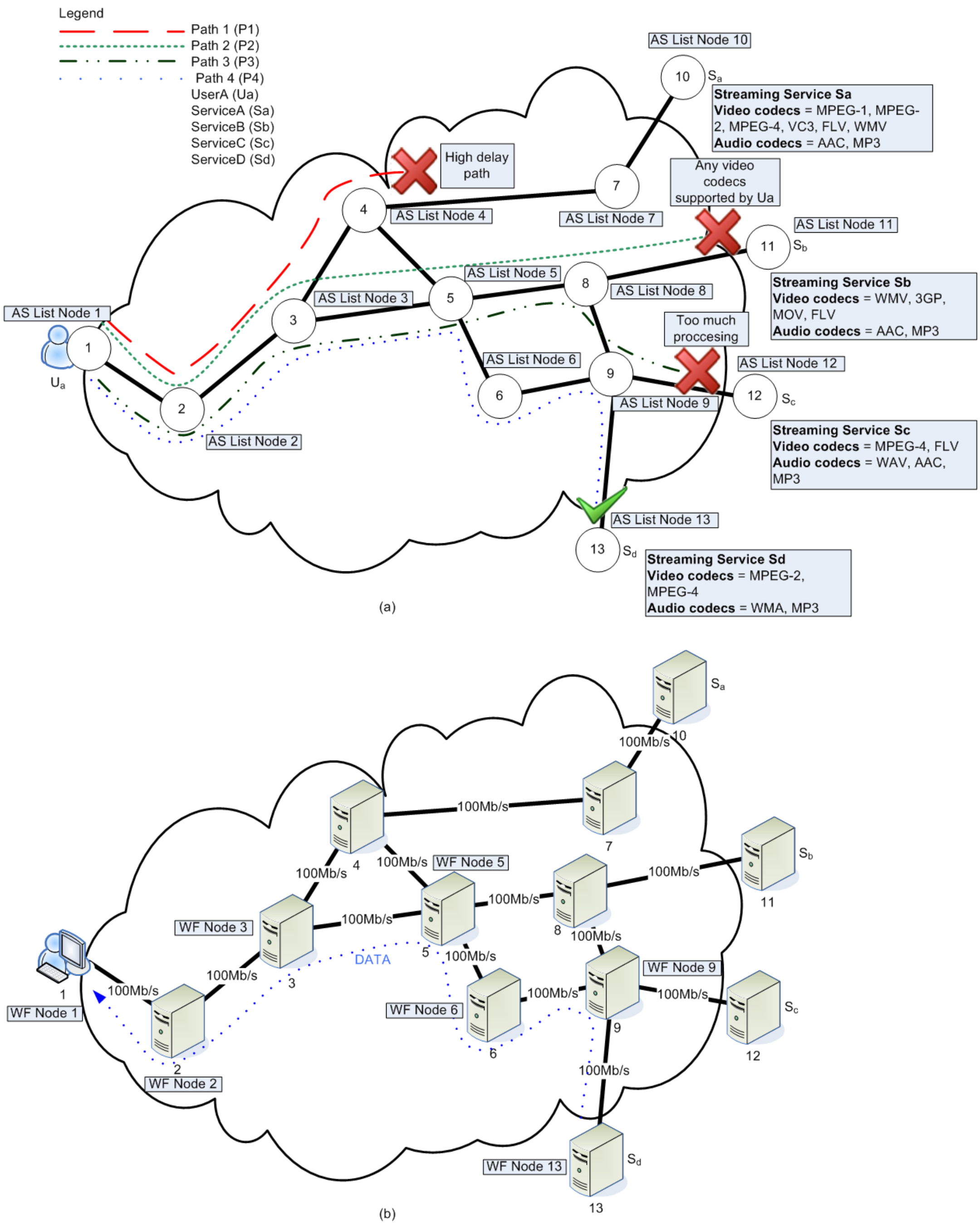


Figure 7.8 a) Adapted multimedia communication use case and b) Implemented testbed

7.7 Preliminary Results

The time required to establish an end-to-end communication (Te-e) and the resource consumption of the process were measured in our testbed. Concretely, different communication requests, asking for different requirements and network functionalities were tested.

Table 7-5 specifies the high-level communication goals for each test.

Table 7-5 Scenario specification

Goals	Scenario 1 (S1)	Scenario 2 (S2)	Scenario 3 (S3)	Scenario 4 (S4)
data_transmission	RN	RN	RN	RN
data_forwarding	IN	IN	IN	IN
data_reception	ESN	ESN	ESN	ESN
decoding	RN	RN	RN	RN
encoding	ESN	ESN	ESN	ESN
security	-	RN,ESN	-	RN,ESN
reliability	-	-	RN,ESN	RN,ESN

In practice, these goal effects are associated to different ASs or combinations of them. Additionally, each AS can be implemented by different AMs. As an example, imagine we have an *encoding* goal which can be provided by *video_coding* and *audio_coding* ASs. Each AS can be then provided by AMs like: MPEG-1, MPEG-2, MP3, WMA, etc. Regarding performance parameters, the average total consumption during the process of negotiation is shown in Table 7-6.

Table 7-6 Nodes resources consumption

Node Type	CPU	RAM
Requester Node (RN)	13%	224,30 KB
Intermediate Node (IN)	5%	113,46 KB
End Service Node (ESN)	9%	192,37 KB

The time required for negotiating the end-to-end communication (Te-e) in each scenario is shown in Figure 7.9. Concretely, in this figure we show the results obtained for the longest path (P4) of our testbed.

Note that in our tests, Tpropagation is almost negligible as we use dedicated links in a local testbed. Tforwarding is also constant as all the involved nodes perform a lookup into its local databases of ASs, and all of them have the same size. Regarding Tresponse, it is slightly different in the ESN than in the IN as it has to insert more data into the Cresp. In this case, each node inserts into the Cresp information about the AS, AMs and QoS parameters that the node can offer.

The gathered results are preliminary results for the different scenarios introduced in Table 7-5. The most representative value is the time required for negotiating the services that will be used (Te-e) and the most influential parameter is the composition time needed to decide which services to use (Tcomposition). Composition can be a very demanding process if full flexibility and the best possible decisions are to be achieved. Mostly, its value depends on the number of goals to successfully target and the number of ASs and AMs supported by a node. As specified in section 7.4, the proposed composition algorithm must calculate all the possible combinations between ASs in order to select the best ones. The more services, the more combinations must be calculated. However, in future work, some techniques for improving this process will be studied. Even though, once a composition is performed, the resulting workflow of services could be stored for future reuse so as to avoid calculating again all the combinations of services.

Finally, note that in the presented prototype, monitoring functions are not implemented. An efficient monitoring system that provides context information is especially important for the development of this solution. In the future, we expect to be able to implement specific monitoring mechanisms to get real context data.

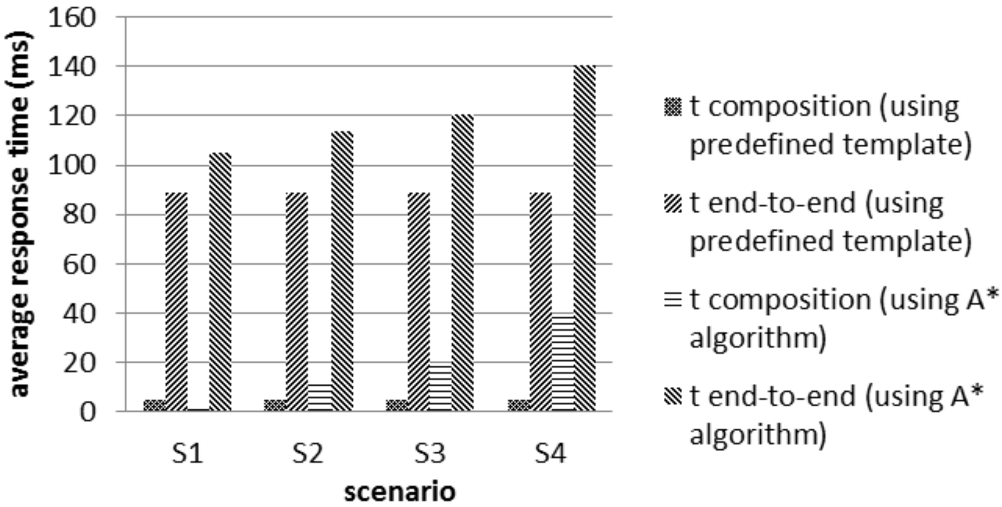


Figure 7.9 End-to-End time and composition time

Chapter 8 Future Internet Service Provisioning

Thanks to the architecture fundamentals introduced in Chapter 7, we now introduce how services can be provided in the Future Internet specially focusing in two main aspects. The first one introduces how SOA paradigm can be applied to context-aware multimedia communications. In addition, a scoring function for selecting different service implementations is presented and particularized for a case of selecting transcoding functions taking into account different video/audio quality assessment metrics. In this case, we focus on specific video transcoding functions, however, other functions like the ones introduced in section Chapter 6 could be added in order to create intelligent networks able to adapt to context conditions and specific requirements. The second aspect is Future Internet economics. This PhD. Thesis proposes costing framework for dealing with socioeconomic parameters within Future Internet architectures. In this context it is necessary to introduce a framework for cost and price that enables requesters and providers to interact and create new business models for the Future Internet.

The solutions presented here were published in [GP2], [GP7] and [GP8].

8.1 Context-aware Multimedia Service Composition using Quality Assessment

There are emerging services, which offer contents in multiple ways and different formats. Popularity of video services such as Youtube or Hulu, have made video traffic in the Internet the most present one. Moreover, the heterogeneity of devices connected to the network is also rising (e.g. handheld, PC, TV, etc.) and it usually requires the creation or the adaptation of services and resources specifically for each target platform. This situation leads to generic and static systems that do not provide the content adapted to the final device or too complex systems that require a big effort in development and maintenance tasks.

In this scenario, the necessity of context-aware systems appears. Their goal is to offer services adapted to the context of users whilst maximizing the Quality of Service (QoS) and Quality of Experience (QoE) of the user and allowing a more efficient usage of resources. However, their deployment requires precise and efficient monitoring and data management systems. One promising approach to efficiently provide context-aware services is using Service-Oriented Architectures (SOA), which divides Services in simpler ones and couples only those that are required or preferred for a specific context.

Solutions based on this type of architectures provide some clear benefits: loose coupling, implementation neutrality, flexible configurability, granularity, task distribution, energy efficiency, efficient use of resources, etc. The division of services could be done based on different aspects such as location, capabilities, functionalities, etc. As said, in this work we follow a RBA decomposition approach where complex services are divided into indivisible or atomic functionalities. Examples of these functionalities are: encoding, acknowledgment or retransmission. Additionally, no matter how the decomposition is done, there are many approaches or visions of doing the composition process. However, there are some points to be taken into account and some common problems that appear in most of current techniques:

- Considering services as self-contained, self-describing, modular applications that can be published, located, and invoked across the network, service composition process can be defined as the combination of those services required to create new processes and services.

- Fulfillment of preconditions when a service that can provide the desired effects exists [111].

- Generation of several effects. It is possible that a service request is associated to multiple effects that can be satisfied by different services.

- Knowledge and context data acquisition and management.

These problems denote a close relationship between optimal compositions and context-awareness. Thus, how to obtain and analyze the context is an important issue, in order to provide a good ground for the service composition process. In this framework, we review different methods that could be applied to evaluate the quality of multimedia services and how to apply multimedia quality assessment to enhance a multimedia service composition process. Context-awareness features are especially relevant, as the ubiquity of mobile devices and the proliferation of wireless networks is enabling permanent access to the Internet at all times and all places. The next step to an Internet of Services is an Internet of context-aware Services.

According to the definition provided by Dey in [104], context is any information that can be used to characterize the situation of an entity, where an entity is a person, place, or object that is considered relevant to the interaction between a user and an application, including the user and applications themselves. Context-awareness refers to the capability of an application or service to be aware of its physical environment or situation and responding intelligently (pro-actively or reactively) based on such context. So, it is important to compose services and dynamically adapt them according to the context information and changes in order to provide personalized and customized services to users. This should allow improving the QoS and QoE of users while optimizing the usage of network and computational resources. Revising literature related to context-awareness [215][216], we define a consensus classification of the context according (but not limited) to the following:

- **User context**: user characteristics, user location, user preferences, and environmental constraints of the user (e.g. working place, home, etc.).

- **Device context**: type and capability of the device.

- **Service context**: service availability, minimum required QoS level for providing the service, and additional parameters that define specific attributes for a service.

- **System resource context**: CPU, memory, processor, disk, I/O devices, and storage.

- **Network context**: bandwidth, traffic, topology, and other parameters related to network performance.

On the subject of service composition, several approaches can be found in literature tackling service composition [217][218][219][220]. For example, in [219] authors propose a classification system in the form of taxonomy, for semantic web service composition approaches, that could be generalized to the global concept of service composition, and applied to compose services at network level. Unlike other solutions, our approach uses the discovery process to find those services to be composed all along the end-to-end path attending to the requirements specified by the requester and, consequently, assuring a certain level of QoS. Regarding to the quality assessment process, an overview on different metrics and techniques is presented. Multimedia quality metrics can be classified into three groups taking into account in which way a reference signal is needed to measure the quality:

- **Full Reference (FR)**, which compares two entire signals, a reference signal (usually the original one) and a compared signal (usually the coded one).

- **Reduced Reference (RR)**, which only compares some signal characteristics (blocking, blurring, ringing, masking, etc.) previously detected.

- **Non Reference (NR)**, which does not use any reference signal to determine the quality of a signal.

Each of them is used depending on the availability of an undistorted signal (reference signal). The most used metrics are FR (e.g. PSNR), due to its low complexity, but as a drawback, both signals are needed: the original signal and the coded signal. Additionally, methods for quality assessment can also be divided into two categories: objective and subjective. Objective methods aim to mathematically estimate the impairment introduced to media resources during compression or transmission whilst the Subjective ones aid in the statistical analysis of sample ratings generated by humans. In this work we used different objective quality metrics for estimating the quality of a received media resource. For video we used Peak Signal-to-Noise Ratio (PSNR) and Structural Similarity (SSIM) [221], while PSNR, audio SSIM [222] and Perceived Audio Quality (PEAQ) [223]were used for audio quality assessment.

8.1.1 Multimedia Quality Assessment

AMs need to be selected according to several parameters such as performance, quality of service or, in case of multimedia applications and services, parameters such as the perceptual quality. Quality assessment can be used for measuring the quality of multimedia communications. The goal is to select the best AM for each communication, and best means that it can provide the highest possible perceptual quality. This section proposes to use quality metrics for deciding which is the best AM to use when an AS of the type transcoding is used. This AS mainly consists on adapting a content taking into account the context of the requester. We introduce a scoring function that uses the measured objective quality and the compression ratio provided by different codecs. However, other parameters can be added, such as performance ones (e.g. CPU, energy consumption).

8.1.2 Multimedia Quality Analyzer

We have developed a multimedia quality analyzer module (Figure 8.1) that calculates a score for each codec supported by the multimedia transcoding service. In this case, each codec corresponds to an AM, implementations of the AS named ”transcoding”. We use a FR system that can use the metrics defined in Table 8-1 for determining the obtained quality.

Table 8-1 Used Full Reference Metrics

Media	Metrics
Image	PSNR, SSIM
Video	PSNR, SSIM
Audio	PSNR, SSIM, PEAQ

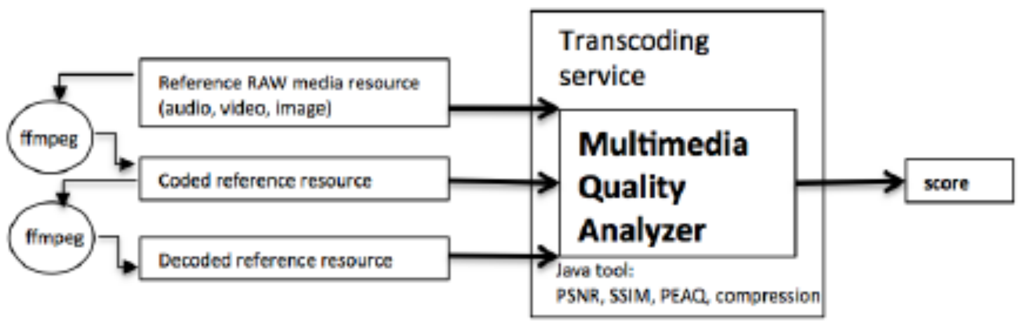


Figure 8.1 Quality analyser module

PSNR is an objective quality metric used to calculate the ratio between the maximum possible power of a signal (in this case an audio or video stream) and the power of the corrupting noise. It is commonly used to calculate the effect of losses in a video or audio signal. SSIM is a metric for calculating the similarity between two images, which lies in the assumption that human visual perception is highly adapted for extracting structural information from a scene. Its application to audio measurement is still being studied. Finally, PEAQ is a standardized algorithm for objectively measuring the perceived audio quality. The inputs of the quality analyzer module are:

1. a media resource (image, video or audio) in raw format.
2. the same media resource coded with a supported codec.
3. the same media resource decoded to raw format.

(1) and (2) are used to evaluate the compression ratio and to obtain the file with losses due to the effect of coding. (3) is used to compare the resulting resource with the original one in raw format (input of the multimedia analyzer) and to measure the differences and impairments. In our system, this process is performed offline, when the system starts or a new codec is added to the system. Then, the system performs all the analyses and stores the results (scores) into a table which is looked up when necessary by the decision-making algorithm.

8.1.3 Score Parameter Definition

The use of lossy codecs allows the compression of the resource size. But, intrinsically, it also reduces the user quality perception. So, it must be found a trade-off of the compression ratio and the perceptual quality. A way to decide which codec is better than another is to consider the perceptual quality and the compression ratio of a coded media resource. Thus, it can be said that a codec is better than other if this presents a better perceptual quality and compression ratio relationship. This can be expressed according to (eq. 11)

$$(eq. 11)\ \ score = (A \cdot perceptual\_quality) + ((1 - A) \cdot compression\_ratio)$$

Where, $0 \leq A \leq 1$. A is a weight, which determines the relevance of each parameter considered in the scoring function. Hence, the relevance of each parameter can be changed. The weight that specifies an equitable relationship between perceptual quality and compression ratio is obtained for A = 0.5. The score parameter is defined in the R set and can take values from -1 to 1:

$$score \in \mathbb{R}, -1 \leq score \leq 1$$

Where -1 and 1 indicates respectively the worst and the best perceptual quality and compression ratio relation. The compression ratio parameter is defined in the R set and it can also take values between -1 and 1:

$$compression\_ratio \in \mathbb{R},$$

$$-1 \leq compression\_ratio \leq 1$$

where -1 indicate that there is no compression between the original and the coded resource, but there has been an increment in the total number of bits, and a positive value (less than 1) indicates a reduction of the total number of bits. The compression ratio parameter mathematical expression is defined in (eq. 12).

$$(eq.12)\ compression\_ratio =$$

$$= \frac{(original\_resource\_num\_of\_bits - coded\_resource\_num\_of\_bits)}{original\_resource\_num\_of\_bits}$$

The perceptual quality parameter can take values between 0 and 1 and it is also defined in the R set:

$$quality \in \mathbb{R}, 0 \leq score \leq 1$$

where 0 indicates, in perceptual quality terms defined by ITU-R in [224], very annoying perceptual quality, and 1 indicates no difference between the original and coded resource. Some of the considered quality metrics do not take values between the defined ranges. So, they must be normalized. The quality metrics to be normalized are:

$$0 \leq PSNR \leq \infty,$$

$$-4 \leq PEAQ \leq 0$$

It is not necessary to normalize the SSIM quality metric as its output range fits into the perceptual quality parameter range. More details are shown in [225].

8.1.4 Combining Audio and Video

The scoring of audiovisual contents should take into account the relationship between audio and video, not only considering their individual scores in an independent manner. The goal is to avoid bad combinations of audio and video profiles, for instance when obtaining combined profiles with very good audio and very poor video (or viceversa).

$$(eq.13)\ score(audioQ, videoQ) =$$

$$= (score_{Audio} + score_{Video}) - \frac{|(A \cdot score_{Video}) - (B \cdot score_{Audio})|}{\sqrt{A^2 + B^2}}$$

In (eq. 13) we define a combined score as the sum of individual qualities and the substraction of each point (videoQ and audioQ) to the line defined by the optimal quality (Ax-By=0). A and B coefficients are defined by the R parameter (eq. 14). This parameter can be introduced by default (system administrator) or by the user as a preference.

$$\text{(eq. 14)}\ A = 0.5 \cdot B = R, R \in [0,0.5]$$

$$A = (1 - R) \cdot B = 0.5, R \in]0.5,1]$$

$$R \in [0,1]$$

Figure 8.2 shows how the quality function looks like for an example value of R=0.66, which corresponds to a 4:3 relation between audio and video. Thus, we give a little bit more priority to video than to audio. However, this relation can be tweaked according to user preferences.

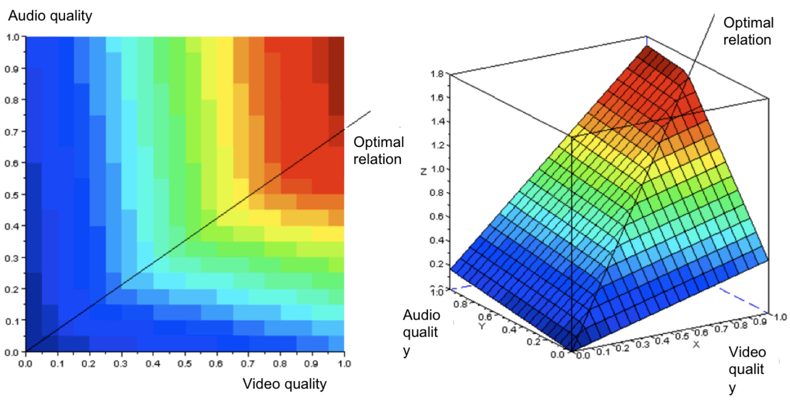


Figure 8.2 Quality function (R=0.66=4/3)

8.1.5 Proof of Concept

We have implemented a quality assessment tool (Multimedia Quality Analyzer) in Java that is able to perform the calculation of the defined score parameter in order to verify its behavior and to test different common codecs against network loss effect. The quality assessment tool is able to use PSNR, SSIM and PEAQ metrics. The testbed scenario is composed of three basic elements:

1. Streaming media server.
2. Streaming client where the media resource is analyzed.
3. Controlled network over which losses are introduced.

The media resource server acts as a video streaming server. It has been used the FFMPEG transcoding software (libavcodec 52.10.0)[226]. This transcoder allows transcoding multimedia resources with a wide range of supported codecs. FFMPEG also supports streaming over a network interface. In our case, we used UDP/RTP to stream the content over the network and to notice the loss effect. In order to remotely control this transcoder a web-service interface which publishes the transcoding service has been deployed. The Analyzer Client is a Java application that realizes requests to the transcoding web-service and receives the streaming sent by the server. Once it receives the coded video resource, it decodes the video and analyzes its perceptual quality. The decoding process is done using the FFMPEG framework too. It is mandatory that the client gets the original resource in raw format to allow the analyzer module to perform the resource analysis. The Controlled Network consists on a PC running the DummyNet [227] network emulator, which permits to emulate networks with a specific bandwidth and Packet Loss Rate (PLR). The analyzed codecs, configuration and input resources are shown in Table 8-2,Table 8-3 and Table 8-4 respectively. It has been chosen these multimedia resources because they are those used in typical quality assessment studies. The packet loss rates applied in the video and audio tests were: 1%, 3%, 5% and 10%. Image analysis considers that if there is a loss in the transmission, all the image is lost.

Table 8-2 Tested codecs (AMs)

Resource	Codec
Image	JPEG, GIF, PNG
Video	MPEG-1, MPEG-2, MPEG-4, H.264, WMV1, WMV2
Audio	MP3, AAC, AC3, Vorbis

Table 8-3 Configuration Parameters

Video	Audio
Bitrate: 1024 Kbps	Bitrate: 128 Kbps
Frame Rate: 25 fps	Sampling frequency: 44100 Hz
GOP size: 12	Bits/sampe: 16 bits
Quantification scale: default	Coding quality: default codec

Table 8-4 Tested Resources

Resource Type	Resource Name
Image	Lena
Video	Foreman
Audio	Vocal quartet, Instrument flute

8.1.6 Results

Results shown in Figure 8.3 are focused on audio because all the mentioned metrics can be used in audio analysis (Table 8-1) and also due to space limitations. A deeper insight on results can be found in [225]. From the PEAQ scoring results it can be concluded that the best audio codec in terms of quality is AC3, followed by AAC. However, in terms of compression ratio the best one is Vorbis, although it is the worst in terms of quality. Thus, if the score parameter is considered, the best scored one is the AC3 codec. The scoring function allows to order different implementations of a specific function, this case a transcoding service implemented by different codecs, in order to determine which is the best one. Moreover, the use of SSIM for audio quality assessment is still being studied.

Notice that this research wants to take special relevance for Future Internet network architectures, which can be used in the development of distributed real-time systems, and will permit the allocation of network services according to each situation and not in a monolithic way. Thus, services must be allocated all along the route, executing just the desired service at each hop, section of hops or end-to-end. Hence, this research should help to pave the way to highly flexible networks, by efficiently applying service-oriented approach to networking, resource optimization and service composition.

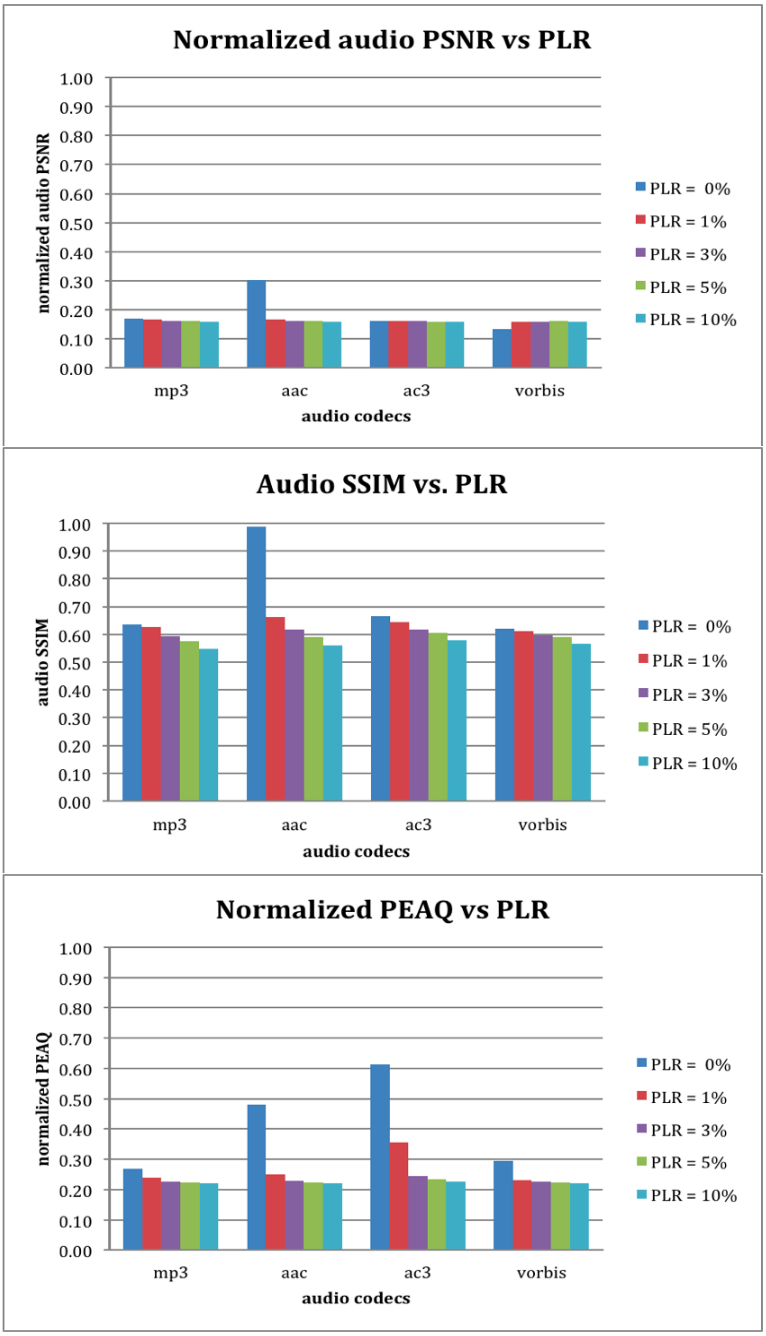


Figure 8.3 Testbed results

Concretely, we only present a proof of concept of the proposed framework, where we particularize a general expression for scoring services in the case of selecting a multimedia codec taking into consideration quality assessment metrics. However, it is important to notice that other parameters can be added to the scoring function. Additionally, each AS can propose a specific scoring function in order to select the best AM that is able to provide it. The scoring of an AMs can be done by each node in the network or, if the profile information for all the nodes is available in the network, it can be done by specific external nodes that can carry out this task. Current services using these techniques will be able to adapt content taking into account the context of users intrinsically. Moreover, the study presented here introduces a way of enabling context-aware communications in the context of Future Internet architectures based on services. Thanks to these architectures new functionalities can be added in an easy and flexible way, allowing the proliferation of new applications while adapting architectures to past, present and newcoming requirements. Regarding to streaming services, advanced video coding techniques such as MDC (Multiple Description Coding), SVC (Scalable Video Coding) or MVC (Multiview Video Coding) for next 3D formats could be placed in the network, and instantiated only if required, to enable transparent media aware networks and save network resources consumption.

8.2 Costing Framework for the Future Internet

This dynamic scenario where services can be published by different providers, discovered and consumed considering the needs an interests of each user, opens the door to new business models based on what the requester is looking for. Services are valuable pieces to be composed and reused. Thus, it is necessary an economical model for providing services to requesters, trying to maximize their satisfaction. It is important to remark that the term 'requester' is not only limited to final users. It covers a wider scope. A requester can be any entity able to ask for a service, including scenarios where a requester of a service is mainly a provider of other services or, even, other services connecting to other services. In the same sense, a user could offer its own services in a transparent way and, thus, it can become a provider.

Moreover, new trends on socioeconomic aspects for the FI are appearing. New business opportunities, emerging communication paradigms (e.g. Internet of Things, Internet of Services, etc.), regulation or innovative uses of Internet are just some examples of them [230][235]. However, newcoming proposals converge in the convenience to meet requester needs, desired QoS, and the price that users want to pay for a service

This section introduces a simple costing framework that may be used in this scenario for pricing services and, thus, dealing with them in a Future Internet scenario. The simplicity of the proposed model is meant to enable its feasibility and fast operation in practical communication systems. Then, it introduces how to use this framework taking advantage of service discovery and service composition processes for providing personalized and adapted services according to context in Future Networks. This framework permits to calculate and compare the cost of consuming a service provided by a set of candidate nodes in the network (even located in different paths to the End Service).. However, it does not imply that the final price is directly this cost; it could also be used by service providers to calculate the network cost of offering some services to consumers and calculate a fee according to different pricing plans (flat-rate, usage-based, etc.) for each consumer, or a set of them.

8.2.1 Costing Framework Overview

Future Internet Socio Economics (FISE) consider a number of research approaches such as: *business innovation, connectivity, confidentiality, costing, drivers, governance, information delivery, emerging communication paradigms (e.g. Internet of Things, Internet of Services, etc.), infrastructure (e.g. network virtualization), new business, overlay applications, security, Quality of Interconnection (QoI), regulation, service provisioning, social welfare or Internet usage* [228][232][234][235][8]. All these research lines converge in the need to meet requester needs, based on its preferences according to requested data, with a desired level of QoS, and the amount of money a user wants to pay for a concrete service or content. This approach gives to users enough freedom to choose among different services and contents in order to fullfill specific technical and non-technical goals while maximizing the satisfaction of users.

Service Composition requires a number of services in order to accomplish the requester requirements. Besides technical aspects, we consider that every AS in a CS has a cost, which also entails non-technical arrangements derived from Socio Economical aspects (e.g. price, value, hypothesis or business models). Thus, the service provisioning process carries a socioeconomic implication to be addressed between providers and requesters.

Costs of services in the network are related to expenses that can be derived from production, infrastructure amortization, energy consumption, research, management, etc. All these expenses have a result that affects the level of Quality of Service (QoS), which is considered as a function of functional and non-functional service quality attributes, such as service metering and cost, performance metrics, security attributes, reliability, scalability, and availability [233]. This has consequences in QoS parameters such as dedicated bandwidth, controlled jitter and latency, and improved loss characteristics [229].

The price of services also depends on the set of assumptions made by providers in order to obtain a balance between cost and benefit. That is the Business Model (BM) that takes into account a series of components like price, convenience, commodity-plus, experience, organization planning, distribution channels, intermediaries, trust chains, and innovation [231]. Depending on provider interests in social and/or economic terms, it will make a hypothesis to establish a value factor in order to set a price on each of the components.

Many assumptions can be considered to make an hypothesis. Some examples are: level of use of the service and network, agreements between networks (e.g. two or more networks may be considered as a unique network), implementation costs of each network, management and maintenance costs of the network, regulation, etc. All these criteria are used to define a dynamic function to calculate a price for each composed service.

To explain the operation of the proposed framework, an Internet model where End Services can be offered by different providers in different domains and can be reached by different paths is used. This model (Figure 8.4) is different from current Internet hierarchical environment. However, different models can coexist in the future. Thus, FI architectures should guarantee compatibility among them and also backward compatibility with current Internet.

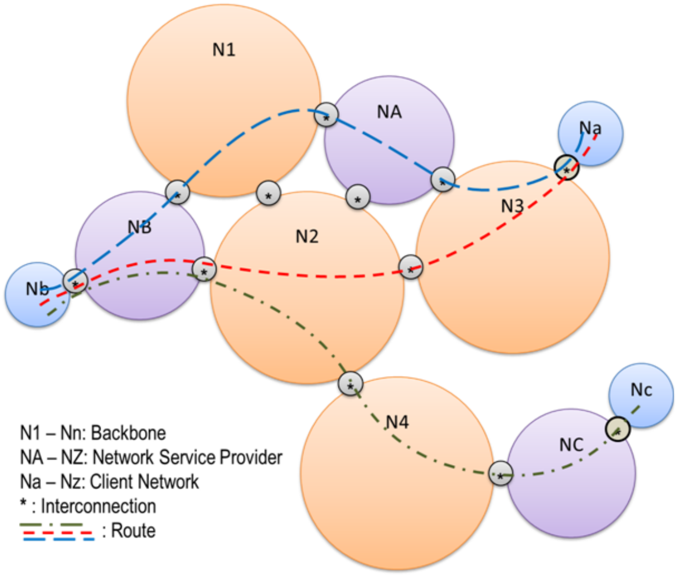


Figure 8.4. The ESN can be reached via different routes

In this scenario, the requester is capable of searching an end service over the network and choosing a combination of available ASs along the path. Service discovery process for an end service will return as a result a set of routes compliant with requester requirements. Each route has different characteristics and, consequently, different costs.

It is necessary to define a socioeconomic model suitable for the proposed scenario. Therefore, in this work, a framework able to calculate the cost for a composed service is introduced. Based on service composition mechanics and granularity, this costing framework is divided into three main components: Atomic Cost, Node Cost, and Composed Service Cost (shown in Figure 8.5).

- **Atomic Cost** (W_{AS}) refers to the cost of an Atomic Service (eq. 15). It is an objective metric, which can be calculated according to the resources consumed by an AS into a node. The cost directly depends on the effect of the AS into the Quality of Service (QoS) parameters associated to its execution. k_R is a constant value relative to each AS.

$$(eq.15)\ W_{AS} = k_R$$

- **Node Cost** (W_N) depends on the cost of the set of ASs (eq. 16) executed in a node. In addition, it depends on a subjective parameter (S_Q) which adds value to the execution of a specific AS. S_Q is a value obtained from a specific function for each AS and depends on the desired subjective metrics, for instance, the relative cost of demanding a specific resource at a precise time. The product of W_{AS} and S_Q for each AS is called ASCost.

$$(\text{eq. }16)\ W_N = \sum_{i=1}^{n} [W_{AS_i} \cdot S_{Q_i}] = \sum_{i=1}^{n} [ASCost_i]$$

- **Composed Service Cost** (W_{CS}) is the aggregated cost of each contributing node (W_N) in a route (eq. 17). From the network service provider point of view, it should be interesting to identify the cost of its network (W_{CS_Nx}) in providing a CS. It could be calculated as the aggregation of costs related to its internal nodes. However, RN, when demanding a service, will obtain W_{CS} as the addition of all WN participating in the CS. It is the same as the sum of the W_{CS_Nw} of each provider in a route from RN to ESN.

$$(eq.17)\ W_{CS} = \sum_{i=1}^{n} [W_{N_i}]$$

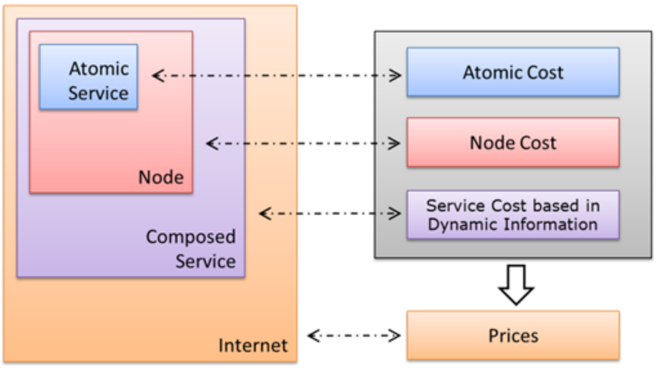


Figure 8.5. Costing framework

Note that we are defining a cost framework focused on atomic services. This fine granularity is useful for creating a dynamic protocol stack, consequence of selecting those required services to establish a communication. However, it is also possible to define a coarser granularity to apply the cost parameter to sets of services or CSs, which are used very often together. Hence, avoiding to associate a cost to tiny individual atomic services that in a practical system are very difficult to quantify in pricing terms. For example, for a node performing forwarding of packets in a network (e.g. a router), the cost parameter can be applied to the combination of the services involved in this operation: encoding of data signals to transmit over a network, data framing and data forwarding.

Thanks to this framework, services would be able to be combined according to the requester needs while allowing the providers to specify how services are charged according to different pricing models. It is important to remark that the costing framework should maintain its simplicity in order to be feasible and implementable. Thus, being able to run in real-time nodes.

Regarding to service composition process, unlike other solutions, presented approach defines a preliminary discovery process that finds those services to be composed all along the end-to-end path considering the requirements specified by the requester and assuring a certain level of QoS.

Furthermore, note that the composition process defined is complex in computational terms. In order to avoid the calculation of the combinations and scores every time a service is requested, past compositions can be stored and reused. Thus, in static environments only new or rare services will be created. On the contrary, very dynamic environments will make this method computationally heavy and not suitable for nodes with low processing capacities.

Service composition is foreseen to be a key enabler for creating flexible and scalable architectures for the Future Internet. Thus, in this context, it is fundamental to create a suitable cost model for dealing with services and to create a powerful environment to develop new business models.

Part IV: Conclusions and Future Work

Chapter 9 Conclusions

This PhD. Thesis reports the author's research done in the fields of media streaming and service-oriented Future Internet architectures, basic elements to advance towards a Future Media Internet.

Internet is becoming a huge, heterogeneous and dynamic network that is growing beyond its architectural limits. The scaling up of required bandwidth (mainly motivated by multimedia traffic), the number of communicating nodes/services and network technologies (wirless, mobile, fixed) are leading the current Internet to a load and architectural crisis which in turn makes it difficult to provide services efficiently and adapted to the requirements and context conditions of users/requesters. Even more, the end-to-end argument and layered architecture design principles of the current Internet are no longer valid to face this situation. Consequently, the Future Internet design should address the shortcomings of current TCP/IP-based Internet architecture.

Multimedia streaming applications are the most hungry-bandwidth applications and, fundamentally, current Internet traffic is video. This trend is expected to grow in the future with the constant appearance of multimedia capable devices, digital content creation and improvement in network technologies. Thus, Internet is evolving towards a Future Media Internet, motivated by users demands asking for video and multimedia contents everywhere. In order to allow Future Media Internet to provide the best possible quality of experience to users, content adaptation, personalization and seamless delivery are crucial issues. However, to achieve this, we need to find efficient solutions for media distribution and adaptation in high heterogeneous and dynamic environments. Error propagation and packet loss take place frequently over today IP networks, even more if considering networks without infrastructure support like mobile, vehicular, sensor networks, etc. These effects can considerably reduce the video quality at the receiver nodes. According to this, handling packet loss and providing error resilience are decisive in streaming applications. As seen in This PhD Thesis we made studies and contributions in challenging environments such as P2P dynamic networks and VANETs. These scenarios were excellent to demonstrate the benefits of the different robust, scalable and reliable proposed mechanisms (MDC, SVC, NC, FEC) for media delivery.

In this work we worked and validated on how to take advantage of synergies among different techniques for facing network/device heterogeneity and dynamism. We explored different streaming systems and have provided solutions in necessary areas covering media distribution, media coding, media adaptation and media signalling. More specifically, we covered device capabilities recognition, delivery context characterization, content adaptation, context acquisition mechanisms, content negotiation, and application layer distribution protocols optimization. Solutions on these areas allowed improving media streaming in challenging scenarios such as the dynamic and heterogeneous peer-to-peer networks or, even, vehicular ad-hoc networks.

In this research we explored source coding techniques (mainly MDC and SVC). Understanding them helps to choose adequate techniques that will make able to optimize resource usage and to improve substantially the overall distribution system quality and performance. These schemes allow distributing media resources in heterogeneous networks, where many different kind of devices are connected (from full-featured desktop PCs to a limited mobile devices such as a PDA). Every client connected to the network can have different features or capabilities (two of the most common ones are available bandwidth depending to the network access and display resolution).

When applying these kinds of techniques, the user experience is not only improved in terms of better resolution at the receiver. The user experience is improved when talking about continuity at viewing time too. Suppose there are lots of losses in the network and the user is displaying a media resource sent via a P2P application. At a given instant, it can stop receiving one out of many descriptors/layers but the resource can continue being displayed although at less resolution (less quality, but the reception continues). Once the network conditions are restored, and more descriptors can be received, the resource can add the new descriptors received in order to improve the resolution. This means, better quality. Note that MDC and SVC approaches behave differently in that situation. MDC offers completely independence among descriptors, that is, just one descriptor is necessary to start receiving a valid bitstream to be displayed. In contrast, the dependency among layers introduced by SVC adds a specific constraint that consists on the need of receiving layers in a specific order, thus, first it must be received the base layer (minimum required) and, then, the successive dependent enhancement layers. The use of one technique or another depends on the capabilities of the receivers and the final application goal and constraints (e.g. storage capacity).

MDC and SVC provide very interesting features such as scalability and error resiliency. However, the main disadvantage of these techniques is that they introduce a new processing stage before transmitting and this operation can be expensive in terms of processing cost and, consequently, extra-delay can be added at transmission time (there must be a trade-off between processing cost and the overall introduced delay). This can be a critical point when talking about a live streaming service, specially when interactivity is required. Nevertheless, the choice between the two techniques will be determined by the specific application or service considered and according to specific requirements.

Moreover, in this work we also studied the performance of flexible MDC systems with incentive-based mechanisms for redistribution. In order to carry out this evaluation, a test-bed has been developed to allow the simulation of MDC systems and the use of incentives. The gathered results show that the proposed solution clearly improves the considered metrics and the overall behaviour of the system compared to the performance of the reference system. These results can be used as reference or guideline for further developments. As an additional outcome of this work we have developed a simulation software which is a valid test-bed that can be used for future studies (based on P2PTVSim). In order to provide a more comprehensive validation of this system other measurements and simulations should be performed to complement the results of this work. More specifically, the measure of the overhead introduced by the use of MDC should be considered in order to see if it is worth deploying these kind of techniques, and trying to define the optimal overhead to be introduced (trade-off between number of descriptors and overhead). Also PSNR quality measurements could be performed by selecting a specific MDC technique and running the corresponding simulations. To complete these results, the effect of churn, concretely the impact of flash-crowds, is desired to be evaluated in order to see if the proposed solution performs satisfactorily under these conditions. Taking into account the obtained results, another line of work that we are starting is the use of MDC for systems or applications with low star-up delay requirements and the corresponding approaches to it.

In conclusion, source-coding techniques are suitable for distributed delivery networks that exploit the benefits of path and terminal diversity to face network loss. Moreover, they are promising techniques to be applied with more revolutionary approaches such as network coding. As introduced in this research, we applied NC operations implemented above the MAC layer while MDC operations are embedded at the application layer in a VANET streaming scenario. Thus, we proposed a joint source-network-coding mechanism assuming a cross-layer implementation. The MDC descriptions are adapted with NC generations. This two stage coding mechanism increases the robustness of video streaming in such networks. Moreover, the NC redundancy is adjusted using a fuzzy inference system controller. Then, the tuning of the amount of redundancy was based on vehicular traffic density and SNR of the communication channel. NS-2 simulations were successfully carried and showed significant gains in terms of average video quality, application layer throughput and average packet loss rate. The video streaming with NC has been compared to the proposed joint coding approach and the proposed scheme gives better performance.

As said, the proposed system also used fuzzy logic mechanisms to adapt the quantity of redundant data that must be introduced in the transmission system in order to improve the network transmission robustness. This mechanism was also integrated in an adaptive beaconing rate control approach called ABR [GP5] to tune the frequency of beaconing rate in response to vehicular traffic characteristics (APPENDIX V). This adaptive feature of the ABR approach makes it suitable for rapid arrival and departure characteristics of vehicular networks (sparse and dense scenarios). Simulations using a realistic city scenario have shown that the ABR approach—in contrast to a fixed beaconing scheme—compromises between beaconing load and cooperative awareness in different vehicular densities and emergency ratios. That is, we also showed that beaconing load is reduced on the cost of cooperative awareness between vehicles, if channel error is considered.

In addition, in this thesis we explored other mechanisms to provide robustness and reliability to video streaming communications. Specifically, we proposed a FEC mechanisms that can be adapted according to the information contained in the RTCP reports exchanged between the source and the destination. We contributed in the proposal of an error resilient scheme based on packet level FEC and packet interleaving technique for reliable video geocasting over VANET. The proposed scheme has used RTCP feedback to optimize FEC redundancy and packet interleaving level as a function of channel variations in a vehicular traffic condition (packet loss rate indicated in the RTCP report). The source vehicle selects (in the geocast region) the RTCP report of the farthest receiver vehicle. This is because longer distance yields higher packet loss, hence lower perceived video quality, thus selecting RTCP of farthest node could recover largest amount of packet loss of the nodes in geocast region. Therefore, we achieved the improvement of video quality and protection efficiency whilst respecting transmission constraints. NS-2 extensive simulations were successfully carried out for significant gains in terms of average video quality and packet recovery. The video transmission without FEC or with fixed amount of FEC has been compared to the proposed error resilient mechanism and the proposed scheme gives better performance. This technique can be used in combination with the above mentioned (MDC, SVC) to provide extra robustness in the packet delivery process.

However, our contributions work mainly at application level. This is the easiest way to introduce solutions to the current Internet as it imposes a rigid and monolithic layered architecture that was introduced assuming the original end-to-end argument where network intelligence was only place at the edges and the network role was focused on data transport. Another way of introducing more efficient solutions is by complex and particular cross-layer implementations. However their main drawbacks are complexity and the fact that they cannot be reused. Sub-layering is another possibility, implementing specific functionalities but limiting their global deployment. Thus, in this situation, we need a shift from a network following the end-to-end argument, building a new network architecture that provides more intelligence to the network side whilst leaving decision-making processes to the edges. Such a shift is required to combat heterogeneity and to be able to create intelligent networks that can provide all kind of services in an adapted manner according to context conditions and the requirements of the different stakeholders of the Future Internet. All the proposed mechanisms (e.g. MDC, SVC, NC and FEC operations), existing techniques and future functionalities could be represented as services to be discovered and composed according to specific communication and users goals. This new flexible and adaptable architecture is fundamental to achieve real seamless media provisioning and, hence, to go forward towards a Future Media Internet that meets the expectations of users and all kind of stakeholders.

The Service-Oriented Architecture (SOA) paradigm offers fundamental design principles that can guide the Future Internet development towards a more flexible and scalable Internet based on functional pieces called services that can be combined according to requesters' needs. In our work, we propose to take as a base a clean-slate and service-oriented architecture (TARIFA project) that focuses on services combination and adaptation to context conditions by means of three main processes: service abstraction, service discovery and service composition. These processes are necessary to enable Future Internet service provisioning in an adapted manner, satisfying the specific requirements demanded by the communication requester and, additionally, maximizing the QoE and making an efficient usage of network resources.

To achieve this, one important feature of the proposed solution is that routing functions are integrated into a semantic service discovery protocol that evaluates context conditions hop-by-hop when a communication is requested. In addition to this, we also propose a service negotiation protocol which would enable us to find and compose services to meet requesters' requirements in an efficient manner. This guarantees the demanded QoS whilst providing appropriate QoE to service requesters. Furthermore, as the research community contemplates clean-slate designs for Future Internet architectures, models and implementations for assessing its development viability of new network architectures are especially needed. In this paper, we provide the main details of a first implementation of the proposed service-oriented solution and discuss the preliminary results we have gathered. These results establish the grounds on which the proposed clean-slate architecture will undeniably enable a more flexible, scalable and dynamic Future Internet.

New functionalities can be added in an easy and flexible way, allowing the proliferation of new applications while adapting architectures to past, present and future requirements. Concretely, regarding streaming services, advanced video coding techniques (e.g. MDC, SVC or MVC for next 3D formats), network transmission operations (e.g. Network coding, FEC coding), video adaptation (e.g. using quality assessment tools), protocol operations (e.g. publish, subscribe, P2P distribution), etc. could be natively placed in the network, and instantiated only if required, to enable transparent media-aware networks and save network resources. In addition, networks will be able to adapt according real data gathered from the network (loss, delay) by means of using quality assessment tools, decision algorithms such as the fuzzy-based algorithm proposed in this thesis, etc. and avoiding implementing more complex methods at higher levels (application).

The architectural concepts introduced in this research does not suppose yet another clean-slate approach. It presents a radical view of the Future Internet, where the necessary functionalities for accomplishing communications, in user devices and in the network and at all levels are considered as services. Services are not fixed but dynamically composed where and when necessary, with respect to user service requirements, network transfer capabilities and surrounding context in the user and the network environments.

Composition of basic network-level services calls for a clean-slate approach to the Internet, while composition of higher level (transport and application) services prompts for an evolutionary approach. Nevertheless, composition of communication services manifests itself as a revolutionary way of looking communications and building communication systems. Thus, the concepts introduced in this work can be introduced gradually in current networks and will allow preparing the road to more revolutionary changes at the network level.

There are many algorithms that can be used for service composition; they have been widely explored in the Web Service domain (application level) to create complex services and mashups. However, in the Future Network we will need to evaluate which is the most appropriate one that can be used according to the kind of services to be composed and desired communication. Depending on the level where the service is (application, transport or network), different constraints should be considered. Composing network services should be fast enough to allow establishing communications, not only between two nodes in a domain, but also in a multidomain scenario. Thus, delay is a critical parameter to take into consideration. In addition, involved devices will present strict processing limitations (routers). On the contrary, we can find application services that can benefit from more powerful processing capabilities that will allow creating more complex and flexible compositions. It remains as future work to explore the requirements introduced at each level. In so doing, we will be able to find guidelines for selecting a composition method or another.

In this PhD Thesis we proposed a composition method based on the combination of services, and the A* algorithm for the final selection. The final result is a set of combined services (chain of services) that will be used to establish the requested communication. Despite the A*-based algorithm was easy to implement in a real prototype and offered great flexibility for selecting between many precalculated combinations of services in different nodes, this algorithm presents some scalability problems and is computational complex. An alternative algorithm or, even, complementary step in the composition process, could be a fuzzy-based algorithm such as the one proposed, simulated and evaluated in this thesis for deciding which parameters have to be adjusted in different environments or which services, mechanisms or already calculated compositions to establish a specific communication.

The last part of this PhD. Thesis introduced two main innovations clearly beyond current state of the art. Firstly, a Service-Oriented framework able to deal with (existing) functionality at all levels (connectivity, transport, application) by considering the provided service and not the technology behind the functionality. All these service functionalities can be seen as services thanks to suitable service-oriented abstractions that allow including existing functionality/protocols as well as new functionality in a flexible way. Secondly, this research presents a novel service-oriented clean-slate architecture generalizing Information-Centric Networking (ICN) approaches. In addition, the service-oriented framework would be the first clean-slate architecture completely aligned with the work done within the ISO WG-7 Future Networks.

Chapter 10 Future Work

Most of the research introduced here has been validated by means of simulators, proof-of-concept prototypes and small testbeds. However, it still remains necessary to demonstrate the feasibility of the proposed solutions in larger and real scenarios. In addition, it would be necessary to compare the performance of the proposed Future Internet mechanisms with current Internet developments in order to identify the benefits and required capacity to move these proposals forward and schedule a realistic migration. Obviously, it will be difficult to surpass the performance of current solutions in a short-mid period of time, but at least, we are setting the basic guidelines for a more flexible Future Internet able to grow in a clean manner according to new services and applications requirements and users demands. By now, we have attracted the interest of standardization bodies such as the ISO (ISO/IEC JTC1 SC6 WG7) where we are participating in the design of the Future Network, introducing the presented service-oriented architecture principles and the defining service composition problem statement and requirements. Thus, there is still a long way ahead in the standardisation of these issues.

In order to advance towards a Future Internet where seamless service provisioning will come true, it remains necessary to cover future work on service-oriented architectures, service composition and service provisioning. This includes addressing different issues such as:

- Compare and evaluate service composition methods in order to find tradeoffs between parameters in order to achieve efficient mechanisms in different environments, according to context parameters and situations.
    - Optimal service composition requires solving the trade-off between: flexibility, temporal overhead and performance. On the one hand, late composition (like late-binding) enables highest degree of flexibility by taking into account more and more accurate input data. On the other hand, at run-time service composition must be performed in few milliseconds, rarely a few seconds are acceptable. FN architecture will solve this trade-off by splitting the service composition into several epochs (e.g. design time, deployment time, run time, etc.). Complex and long running tasks will be performed “early” while simple and, thus, fast tasks can be performed “late”. Note that service composition also involves the task of selecting a service of the next lower level.
- Explore service validation mechanisms to ensure that the communication will meet user expectations.
    - The service composition task can potentially deal with a huge volume of variables during the decission process according to the required or desired decision-making accuracy and performance. In

addition, it can involve different nodes all along the end-to-end delivery path involving different available services, mechanisms and resources. User expectations depend on service composition outputs as this process has to determine which services best meet user requirements. In order to avoid unsatisfactory user experience, the service composition output should be validated to ensure that the selected services meet users' expectations and, in addition, that the network can provide the requested services. This process can be done once a composition is performed and the output is provided (desing-time or run-time). Moreover, service composition validation will evaluate the matching between user requirements, service goals and available resources. The output of the validation process can be a score useful for determining the degree of quality of experience to be achieved.

- Mechanisms to address communications and service discovery across domains in a scalable way. Scalability is a main concern that should be addressed in this kind of information-centric, content-centric and service-oriented architectures.

- Exploring different strategies for service allocation in order to decide which is the best one considering different constraints such as performance, time required for allocation, and, specially, resources reservation.

- Solving intra-domain and inter-domain (multidomain) service discovery and composition.

- Exploration on suitable models and semantics able to describe QoS requirements, available resources and services.

- Exploring suitable semantic addressing schemes for assigning locators and identifiers to services and resources present in the network.

- Introduce automatic mechanisms for establishing agreements between entities when using different services from different stakeholders.

- Explore new business models that exploit the benefits of such a flexible architecture.

- Explore new distribution mechanisms for secure and efficient media delivery, throughput maximization and resources usage optimization. In that sense, network coding supposes a promising research field that can be greatly empowered by the use of mechanisms such as source coding and incentives. In addition, proposing architectures from scratch would open the door to the native adoption of these new functionalities inside the core network in a progressive manner.

- Deploy a more realistic testbed and find interactions with other bigger testbeds/platforms (e.g. German Lab) using user-centric testing (e.g. Living Labs). Living Labs are specially interesting for testing the experience with multimedia applications.

- Empower standardization activities in ISO/IEC JTC1 SC6 WG7, ITU-T SG13, ETSI, IEEE NGSON, etc. and promote collaborations and liaisons between them.

Advances introduced in the basic architecture of the Future Internet will allow optimizing the delivery of multimedia content and will pave the ground towards a Future Media Internet.

Part V: Appendices

APPENDIX I Service Composition for the Future Internet

Service composition can be defined as: composition of those processes required to combine and link existing services (atomic and composite services) to create new processes and services. Service composition is a fundamental part of Service Oriented Computing [57], and is a process that plays a key role in Service Oriented Architectures (SOA [58]).

This section introduces a state of the art on relevant current service composition techniques, frameworks and initiatives.

I.1 Service Oriented Architecture (SOA) Paradigm

SOA paradigm proposes to organize and to use distributed capabilities operating under the control of different ownership domains. Services are the essential building blocks of SOA, and it is based on three premises. First, use of self-contained services and data. Second, use explicit descriptions instead of implicit assumptions. Third, use well defined interfaces between services

In addition, SOA defines three basic elements: Service User, Service Broker and Service Provider. The latter two, create a service oriented communication system. The Service User is composed of an Application and an Application User. To allow the integration and support of different protocols and formats, an Adapter can be placed in order to make the corresponding mappings.

Moreover, this paradigm can be related to Component Based Computing (CBC) [59] design, based on the decomposition of a problem into several pieces called components. Then, these components can be composed in order to fulfill specific problems and covering the requirements needed at specific times. This approach gives important benefits in terms of flexibility, scalability and adaptability, as the components are instantiated as needed.

The benefits of this type of architectures are the following ones: loose coupling, implementation neutrality, flexible configurability, persistence, granularity and task distribution As specified in [57], a SOA system implies the process of discovering and composing the proper services to satisfy a specification, whether it is expressed in terms of a goal graph, a work flow, or other model.

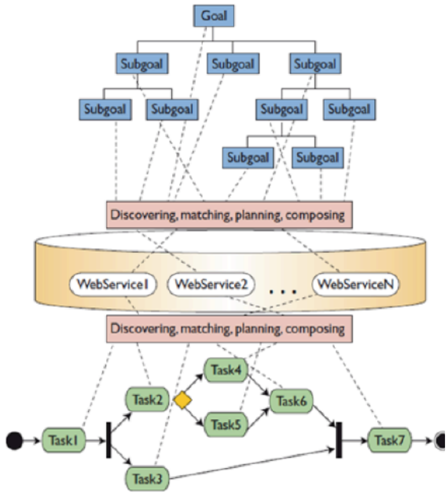


Figure A.1 Service Oriented Architecture [63]

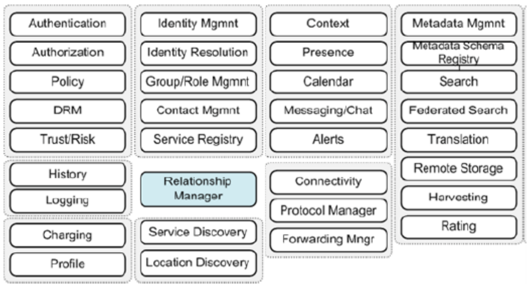


Figure A.2 ROSA set of services [53]

I.2 Services

Services are the fundamental blocks or basic components where service composition process is established. Although they are defined in multiple languages and multiple representations [60][61][62][63][64], all of them maintain certain premises expressed by the following definition: “Services are self-contained, self-describing, modular applications that can be published, located, and invoked across the net”.

In [61], the abstraction of resources as services, presenting a framework named ROSA that works with a set of services (Figure A.1, Figure A.2), is considered. ROSA tries to take a similar principle and framework to Web-Services, but applied to future heterogeneous communication networks to interconnect business borders and administrative domains.

In [64] authors define services as self-contained functions that accept requests and return responses through a well-defined interface. They use logical language (the F-Logic dialect Flora-2/XSB [65]) to represent them, as shown in Figure A.3.

```
fpn:atomicService[
  spec     -> fpnSpec:semanticServiceSpecification[
              conditions ->> fpnCond:condition[
              precondR -> ${N:string, N:parameter, X:person, X[name->N]}:reification,
              precondS ->  "N:string, N:parameter, X:person, X[name->N]":string,
              posEffR  -> ${P:phoneNumber, P:parameter, X[phoneNumber->P]}:reification,
              posEffS  ->  "P:phoneNumber, P:parameter, X[phoneNumber->P]":string]],
  grounding -> fpnBridge:serviceGroundingSpecification[
                 serviceImplRef -> "fpnRef":string,
                 operationName  -> "doIt":string,
                 inParamSeq     ->> {_#:oSP[ord -> 1, str -> "N":string]},
                 outParamSeq    ->> {_#:oSP[ord -> 1, str -> "P":string]}],
  properties -> fpnProps::minServProps[
                 serviceName  *=> fpnSNType:enumeration[type -> string,
                                                 values ->> {"findPhoneNumber":string}],
                 providerName *=> fpnPNType:enumeration[type -> string,
                                                 values ->> {"HPI":string}],
                 cost         *=> zeroCostType,
                 payment      *=> noPaymentType]].
```

Figure A.3 Service Representation using Flora-2/XSB [57]

In [60] authors introduce another approach that tries to cover dynamic software composition within a specific service domain, using existing technologies and without the need for a complex compositional language. In this framework services collaborate to form new composite services (Figure A.4). They also define these composite services, presenting a new common interface that must be constructed at runtime which allows other services to interact with this set of collaborating service components as if it was a single service.

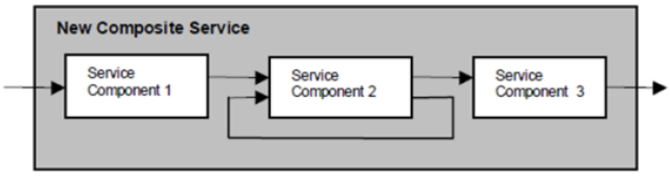


Figure A.4 Composed Service

I.3 Web Services

One of the environments where Service Oriented Architecture (SOA) is widely deployed is in Web Services. Note that the specification of SOA in Web Services (WS-*)[72] includes specifications, standardization and its implementations. Often related to higher-level services, Web Service composition is an extensive subject in the literature.

Web services are self-described software entities, which can be advertised, located, and used across the Internet or a network using a set of standards such as SOAP, WSDL, and UDDI. They are described using WSDL (Web Service Description Language), and they operate jointly using mainly SOAP (Simple Object Access Protocol) and UDDI (Universal Description, Discovery and Integration) standards [59] [73].

Web services encapsulate application functionality and information resources, and make them available through programmatic interfaces, as opposed to the interfaces typically provided by traditional Web applications, which are intended for manual interactions. In literature, it is possible to identify the following main types of Composition:

- Static Composition - services to be composed are decided at design time
- Dynamic Composition - services to be composed are decided at run-time

Some languages for web service composition are the following ones: BPEL4WS (Business Process Execution Language for Web Services), WSFL (Web Services Flow Language), XLANG (BizTalk), BPML (Business Process Modeling Language) or ebXMl BPSS (Business Process Specification Schema). However, these languages are usually used to compose service manually at design time. Currently, Web services are usually described using WSDL descriptions, which provide operational information. WSDL descriptions do not contain (or at least explicate) semantic description, they do specify the structure of message components using XML schema constructs. For that reason, there are many projects (revised in next section) that extend this representation in elements and attributes, using DAML+OIL, OWL-S, or Web Service Modeling Ontology (WSMO) ontologies. The use of ontologies permits to represent Web service descriptions in a machine-interpretable form like DAML-S.

Some requirements introduced by Web Services are the following ones:

- Representation of an abstract Web Process
    - Representing/specifying the abstract process in a proper form
- Discovery and Interoperability of Services
    - Need to manually or automatically search for appropriate services
    - The discovered services should interoperate
- Efficiency of a Composed Web Process
    - Need to compose processes which are efficient in terms of performance
- Process Execution
    - Adopting a suitable technique for executing the composed concrete process
- Process Monitoring
    - Using a monitoring technique for run time analysis of the Web process execution

Although web services operate at the top of current Internet stack, they apply the core concepts we are borrowing in this project. Moreover, composition techniques have been seen in this field. Specifically, in the field of web service composition, we can find available methods for composing services, such as scripting and coordination language [66], rule-based systems [67], planning [68], situation calculus [69], data view integration [70] and integer programming [71] to name a few. They vary in their ability to represent and model non-functional properties of the service, to verify the correctness of the composite service, and to automate the process of service composition.

I.4 Service Composition

In [71] authors propose a classification system (TCS, Taxonomical Classification System) in the form of taxonomy for semantic web service composition approaches. This classification can be applied to the global concept of service composition at all levels. TCS provides a classification providing four levels (except workflow based node) of hierarchy. The focus of classification at first level is the amount of user involvement in the composition process. It provides three nodes Manual/User-Defined, Semi-Automatic, and Automatic. The second level, which is based on the procedure or steps followed in the composition process, creates template and instance based categories under user-defined node while Declarative, Workflow, Template/Instance, AI Planning [75], Ontology based, and Hybrid categories under Automatic node [76]. The third level of taxonomy is based on the amount of dynamicity involved in the composition process and the technology used to implement the procedure adopted in second level. It creates dynamic and static nodes under each of template based, instance based, and workflow based nodes of second level; chaining, logic and rule based under AI planning based and context and agent based under ontology based node of second level [75]. The fourth level of taxonomy produces the categories as shown in Figure A.5 and is focused on some variations possible in the technology adopted in the third level. The fifth level of taxonomy in workflow-based node is based on the centralization level in the execution process.

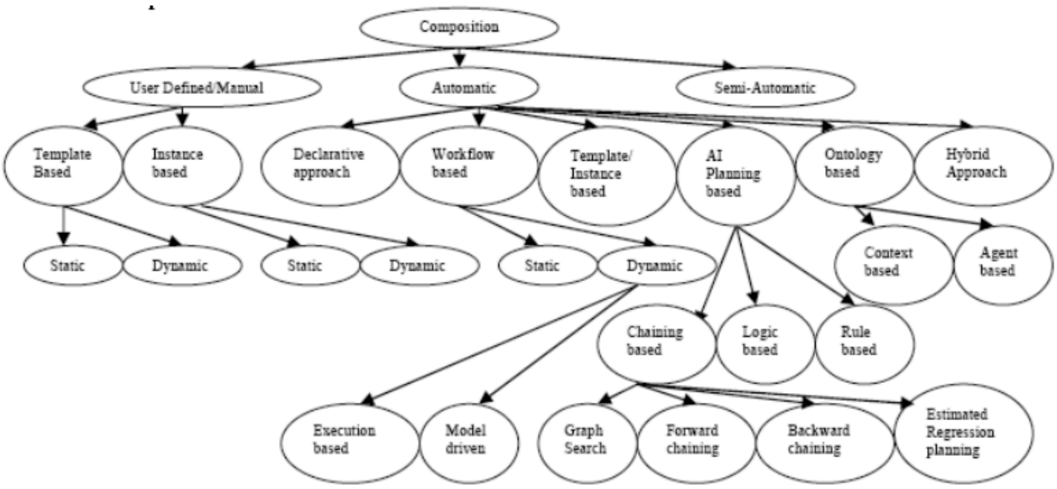


Figure A.5 Service composition taxonomy [63]

These approaches do not plan from scratch like other approaches. They are manually configured or described. The most used methods in this area are template or instance-based compositions, which use recommendations methods to choose a specific pre-designed composition among all available templates or instances.

I.4.1 Template-based and instance-based approaches

This approach usually takes outlines of permissible workflows (or templates) and instantiates one of them to obtain an executable workflow. This process can be based on a set of rules that covers policies, context events and constraints. Policies are specifications of behavior that the composition should have, usually described as Event-Condition-Action (ECA) formulations, while context parameters and constraints are used to describe conditions that affect these policies.

This method could be applied alone, where the initial template should be hand-coded or can work jointly with other automatic methods generalizing the workflows that these automatic approaches extract, in order to re-apply a known workflow for a new situation.

The policies can also guide plan adaptation to external events (e.g. changes in context constraints or the performance of the service) while a composition is being executed.

The template-based approach works best for environments where compositions follow regular patterns (in general more static networks with infrastructure) and its adaptation is required to be along anticipated context. However, the complexity of this approach depends on its generalization. This results on the number of instances that could be selected. Given a template of length λ services, where each instance has at most M possibilities, the worst case of complexity to produce executable workflows is $O(M^{\lambda})$ [77].

Some approaches [78] take this methodology along with templates of services' semantic descriptions, and propose a template-based searching mechanism to allow dynamic searching of business process partners at design or deployment time of a web process.

In [79], a method is presented to compose Web Services by applying logical inference techniques on pre-defined plan templates. The service capabilities are annotated in DAML-S/RDF and then manually translated into Prolog. Now, given a goal description, the logic programming language of Golog [80] (which is implemented over Prolog) is used to instantiate the appropriate plan for composing the Web services. Golog is based on the situation calculus and it supports specification and execution of complex actions in dynamical systems.

Other methods [81] introduce the notion of composition templates only to provide composability soundness; compositions are associated with each composite service and are used to compare values added by different compositions. To check whether a composition is sound or not, stored templates are used to store a composite service's template in the service repository. A service composition is considered as sound if its template is the subset of a stored template.

I.4.2 Declarative Approach

In declarative composition, composite services are generated from a high-level declarative description. The technique uses composability rules to determine whether two services are composable [82]. Most of the time these rules act as constraints that must be satisfied in order to compose a service. The rules are used to generate composition plans that conform to a service requester's specifications. Techniques that fall under this classification usually tend to reach optimality of composition against some defined objectives (e.g., cost, time…etc.) as they are mathematically modeled. Mostly, the optimality can be achieved by mapping rules to constraints and trying to solve them using operation research methods [83].

FUSION is a software infrastructure system that provides the common infrastructure elements needed to support service portals [84]. Given a user service specification, it automatically generates a correct and optimized execution plan, then executes this plan and verifies the result. The most important advantage of this system is its ability to generate an optimal execution plan automatically from the abstract requirements that a user may specify. Also, this composition system verifies that the result of the execution plan meets the user's requirements, and if not, it immediately recovers this execution plan. However, this system creates a bundle of services and uses it to specify an execution plan by choosing services from this bundle. Web services are evolutionarily increasing and determining which subset of these web services to use will limit the choices for the user and will not guarantee the optimality of the result. Also, there will be a probability of using web services that no longer exist.

A technique suggested in [74], which extends the 'inverse rules' query reformulation algorithm to generate a universal integration plan to answer a range of user queries. The inverse rules algorithm unites the inverse rules with the user query to produce a datalog program. A technique was developed to map datalog programs to integration plans that can be executed by a streaming, highly parallel execution engine called Theseus. Moreover, a mediator system is described that accepts a user query and returns a URL of a new dynamically composed web service that can answer a class of user queries similar to the user query. The system has many limitations. First, the system does not support any monitoring mechanisms that test the validation of the composition. This may result in improper execution of composite web services. Second, there is no mechanism to automatically model the newly generated services as a data source for upcoming compositions. Third, this technique lacks semantic representation and composition of these services, thus resulting in an error prone composition.

SELF-SERV [85] is a framework for dynamic and peer-to-peer provisioning of web services. In SELF-SERV, web services are declaratively composed, and the resulting composite services are executed in a decentralized way within a dynamic environment. The framework uses and adapts the state-charts as a visual declarative language. The significant advantage of SELF-SERV is the peer-to-peer service execution model, whereby the responsibility of coordinating the execution of a composite service is distributed across several peer software components called coordinators. Nevertheless, this system does not provide a method to create a composition at runtime for services. Also, it does not consider any semantics of web services during composition decisions. Moreover, this technique imposes some unrealistic requirements that should be implemented by service providers from a partial point-of-view.

Another significant work was performed in [83] to reduce the complexity and time needed to generate and execute a composition and improve its efficiency by selecting the optimal services at the current time. This research proposed an architecture of dynamic web service composition by runtime searching of registries to find services. Therefore, this technique does not use any service template. Moreover, it reduces the dynamic composition of the web services to a constraint satisfaction problem where any linear programming solver can be used to solve it. Also, it insures the optimality of the web services selection based on domain specific QoS parameters identified by the user. However, this approach does not support user interactive participation in the composition process, which is sometimes important to ensure the user's satisfaction.

enTish [86] is somewhat different from typical composition platforms. Services are typically created on the fly to realize client requests. Anyway, most frameworks are based on the assumption that first the business process has to be created. For enTish, a different architecture is needed, since client requests are expressed in a declarative way using formal languages. The declarative approach consists of two phases: the first phase takes an initial situation and the desired goal as starting point and constructs generic plans to reach the goal. The latter one chooses one generic plan, discovers appropriate services and builds a workflow out of them.

The first phase is realized using PDDL (Planning Domain Definition Language) and estimated-regression planning as used in XSRL (XML Web-services Request Language), which must provide machine-readable semantics and specify the abstract service behavior. The second phase may be realized by using existing process modeling languages, such as BPEL.

I.4.3 Artificial Intelligence (AI) Planning

AI Planning is a problem solving technique where knowledge about actions and their consequences is used to identify a sequence of actions, which when applied in a given initial state, satisfy a desired goal [87]. A planner can receive three inputs mainly:

- Initial state: describes the starting state of the application domain. It is normally called world.
- Goal state: describes the desired world state.
- Domain description: describes actions that, when invoked, transform the world states.

The composition engine may be implemented by a number of different composition methodologies. For example, AI Planning has proven to be an effective tool for service composition [88]. Services can be represented in terms of their non-functional and functional properties. Non-functional properties describe service provider details and Quality of Service parameters. Functional properties contain descriptions of service operations in terms of inputs, outputs, preconditions and effects, which makes it easy to translate them into planning actions.

Planning systems need semantics for describing the available actions, thus to describe the formalized domain.

The simplest form of domain formalism is based on state-transition models. The state describes the world at a certain point in time, such as an initial state or goal state. Actions perform transitions between states. An action can be described with a list of preconditions, states and post conditions.

Planning technologies differ in the complexity of the problems they can handle and the representations that they use. Moreover, they employ different search algorithms to synthesize plans and the constraints they observe [89]. We can find different planning technologies:

1. State space planning:
    - Simplest planning algorithm

- Search space is a subset of the state space
- There are two types, based on the search starting point.
  - Forward: chaining planner searches in the space generated by applying to each state all actions whose preconditions are satisfied, starting at the initial state.
  - Backward: search algorithms start at the goal and apply inverses of the planning actions to produce sub-goals, stopping if the initial state is reached. The main limitation of state-based planners is that their performance reduces with the size of the search space.
- To address this performance limitation researchers employ heuristic functions to estimate the usefulness of the alternative actions a planner can choose from. Heuristic methods are found through discovery and observation of the planning process. They guide search algorithms often based on feedback from past executions. This initiated a new type of planning, planning with control knowledge, as discussed later on Heuristic Search Planner (HSP).

2. Plan space planning:
   - Search the space of partially specified possible plans. As a result, the searching process becomes a plan refinement operation.
   - There are two kinds of step that can be taken in constructing a plan:
     - adding an action
     - adding an ordering constraint between actions.
   - This type of planning is called "partial order planning", because until the ordering constraints are added, the order in which actions are taken is not specified. This approach avoids extensive backtracking that slows down a state-space planner.
   - An example is the Universal Conditional Partial Order Planner (UCPOP).
3. Planning graph techniques:
   - Employ graph structures to represent search spaces.
   - Given a problem statement, the planning system explicitly constructs and annotates a compact structure called a planning graph.
     - Represents a plan as a flow of truth values through the graph, which has the property that useful information for constraining the search can quickly be propagated through the graph as it is being built.
   - Graph-based planners then exploit this information in the search for a plan.
4. Hierarchical Task Network Planning:
   - Are based on the notion of hierarchical decomposition, also known as reduction or expansion of an action. This process decomposes an abstract action into a group of steps that form a plan to implement the action.
   - The main objective is to produce a sequence of actions that perform some activity or task. The description of a planning domain consists of a set of actions, as well as a set of methods, which prescribe how to decompose a task into subtasks.
   - The description of a planning problem contains an initial state; however, instead of a goal formula there is a partially ordered set of

tasks to accomplish. Planning progresses as a recursive application of the methods to decompose tasks into smaller and smaller sub-tasks, until primitive tasks, which can be performed directly using planning actions, are reached. For each composite task, the planner selects an applicable method and instantiates it to decompose the task into subtasks. If the plan later turns out to be infeasible, the planner backtracks and tries other applicable methods.

5. Model Based Planning:
   - Planning based on model checking is a methodology that aims to address non-determinism, partial observability and extended goals.
   - It treats the domain as a nondeterministic state-transition system, where an action may have multiple outcomes. Temporal logic formulas are used to express the set of goal states and the conditions of the final plan execution. Planners use a state transition system and a temporal formula to generate a plan that controls the system evolution so that all of the system's behaviors make the temporal formula true.

I.4.4 Ontology based

This technique facilitates the semantic dynamic composition of web services. The ontological descriptions and relationships among web services are used to automatically and semi-automatically compose web services (in this case, any service). The ontology-driven approaches mainly compose the services based on the goal-oriented inferring and planning[90].

The fact that web ontologies are becoming too large to be used in a single application has stimulated many researchers. For instance, a distributed architecture – called Materialized Ontology View Extractor (MOVE) – is introduced in [91] and optimized for the extraction of sub-ontology from a large scale base ontology. This work has been extended in [92] to address the issue of semantic correctness of the resulting sub-ontology. Moreover, large distributed ontology framework for tailoring ontologies in the Grid environment has been investigated [93].

Most of the ontology-driven techniques mark-up web service descriptions with ontologies and develop algorithms to match and annotate WSDL files with relevant ontologies. The possible compositions are obtained by checking the semantic similarities between interfaces of individual services (semantic matching) and considering the service quality (QoS matching). Then, these compositions are ranked and presented according to these two dimensions.

A semantic context-based approach for composing web services is proposed in [94]. The composition is performed based on understanding the semantics of interactions and capabilities of the elementary services. A conceptual architecture that enables ontologies to integrate models, languages, infrastructures, and activities to support reuse and composition of semantic web services is introduced in [95].

To achieve semantic composition, these techniques mostly require a domain-specific ontology design that defines explicit formal specifications of the concepts and relationships among the concepts. It might also require an extraction module that helps us in building ontologies from service profiles. The example shows the taxonomic classification of concepts as well as the relationships that exist between entities.

A dynamic web service composition system is proposed in [96] where a service is not requested and composed by its syntax but by its semantic. To satisfy the requirement for semantic support, the system comprises three sub-systems: Component Service Model with Semantic (CoSMoS), Component Runtime Environment (CoRE), and Semantic Graph based Service Composition (SeGSeC). CoSMoS integrates the semantic information and functional information into a single semantic graph representation. CoRE provides a unified interface to discover and access components implemented in various component technologies to make them interoperable with CoSMoS components. SeGSeC is a semantic-based service composition mechanism that allows users to request a service using a natural language sentence and it generates the execution path. The major contribution is that this work is semantic-based. Moreover, the CoRE component enables the proposed system to interoperate with other legacy existing systems without any updates to these systems. In addition, the composition technique is somehow controllable by the end users. However, the set of queries that can be used to test the system's functionality is quite limited (not to mention that the system was designed for very limited scenarios and it needs a lot of add-ons to be able to satisfy other types of queries). Finally, the system does not have any monitoring tools for executing the composite services. This implies that execution failures are not monitored at runtime.

Another ontology-based dynamic service composition research is presented in [97]. The authors used the Ontology Web Language (OWL) and DAML-S to provide the semantics needed for web service composition. However, semantic description is considered as a part of the process for another sophisticated technique. They have developed a service composition prototype that has two basic components: a composer and an inference engine. The inference engine stores the information about known services in its Knowledge Base (KB) and it has the capability to find matching services. The inference engine is an OWL reasoner built on Prolog. Ontological information is written in DAML and is converted to RDF triples and loaded to the KB. The engine has built-in axioms for OWL inference rules. These axioms are applied to the facts in the KB to find all relevant entailments. The composer is the user interface that handles the communication between the human operator and the engine. The composer lets the user create a workflow of services by presenting the available choices at each step. The advantage of this work is the use of semantic web for web services composition. Yet, this work suffers from centralization, thus yielding scalability and availability problems. Moreover, it requires passing redundant messages between the coordinator and other parties, which causes an inefficient use of the bandwidth. Also, it uses filtering on a set of pre-discovered services, rather than dynamic matchmaking and loading.

I.4.5 Generating Process Templates using Process Mining technologies

Data mining techniques are usually strictly related to process mining problems [98]. In fact, supposing a certain number of tasks within a process, the number of possible different executions is an exponential number (while not all of them have the same probability to be actually executed). Here is where data-mining technology helps.

In [99], while working in the area of software engineering processes analyzed three different methods for process discovery: neural networks, Markovian models, and an algorithmic approach. Their work permits to generate explicit process models, and to measure the actual gap between process model and actual observed behavior [100] introduced the idea of extending process mining to workflow management, by analyzing the events recorded in a log and by identifying constraints. Their approach is based on an algorithmic approach; by enumerating tasks instances and applying a folding procedure E. M. Gold in [101] shows that the problem of finding a state-machine compatible with a set of recorded data is NP-hard. This problem has an analogy with the process-mining problem; even if an important difference is that process mining needs to take into account concurrent tasks. Another important research stream is the one linked with the capability of identifying uncertain variables (either due to low frequency of their observation either due to incomplete datasets). In [102] a probabilistic workflow model and a learning algorithm that is capable to compute in a polynomial time is proposed. This model is based on directed acyclic graphs (DAG) where each node represents a task and the arrows represent dependency relationships, so that given the predecessor tasks, a task execution is independent from the other ones. The learning algorithm builds such graph by connecting the various tasks (nodes) on the basis of their joint instances in a log file.

Finally, other approaches of [103] are based on clustering techniques on data contained in log files. The solution is based on dynamic Bayesian networks or Petri stochastic networks. More in particular, factorial Hidden Markov Models are suitable for modeling parallel tasks. The major limitations of these mining approaches when applied to workflow modeling are: start and conclusion time of instances, and synchronization constraints.

I.5 Context Awareness

Other relevant point to take into account in this research is the context processing. According to the definition provided by Dey in [104]: “Context is any information that can be used to characterize the situation of an entity. An entity is a person, place, or object that is considered relevant to the interaction between a user and an application, including the user and applications themselves”. According to [105][106][107] a consensus classification of the context can be according to the following:

- The user context can include the user characteristics, location, preferences, and environmental constraints (e.g. working place, home, etc.).
- The device context can include the type and the capability of the device.
- The service context can include availability, required QoS level, and performance.
- The system resource context can include CPU, memory, processor, disk, I/O devices, and storage.
- The network context can include bandwidth, traffic, topology, and performance.

In order to manipulate context data, it must be in a format compatible with the models that will be used in context data processing such as reasoning and situation recognition. These models can be object oriented, ontology-based, rule based, logic based, semantic-based or based on fuzzy logic sets.

A key feature when considering service composition is context-awareness [65]. Context-awareness refers to the capability of an application or service to be aware of its physical environment or situation and responding proactively or reactively and intelligently based on such context. Thus, it is important to compose services and dynamically adapt them according to the context information and changes in order to provide personalized and customized services to users. This will allow improving the QoE of users while optimizing the usage of network and computational resources. This aspect is especially interesting as today the ubiquity of mobile devices and the proliferation of wireless networks will allow everyone permanent access to the Internet at all times and all places. It can be said that the next step to an Internet of Services is an Internet of context-aware services.

APPENDIX II Future Internet Architectures description

II.1 SONATE

SONATE [110] approach is based on principles of service oriented architectures (SOA), foreseeing Future Internet architecture as a large, and distributed (software) system, where a set of services communicate, cooperate, and inter-operate with each other. The goal of this service-oriented networking definition is to offer fine-grained functionality to applications to chose from.

II.1.1 Composition Framework

They define services/mechanisms as an abstraction of specific algorithms and data structures used to implement a functionality. Hence, each service provides a self-contained functionality and a well-defined interface that must not take assumptions about internals of other services in order to loose coupling among them.

According to their scope, the services are conditionally grouped into three service clouds:

- **Application Services Cloud.** The application services cloud offers services related to application functionality. Examples of such services are authentication, application notification service, application information exchange service, etc.
- **Mediation Services Cloud.** The scope of the mediation services cloud incorporates services related to network functionality. Typical example of mediation services are connection establishment and release, flow control and congestion control, etc. The presented approach focuses on this cloud.
- **Connection Services Cloud.** The connection services cloud encompasses services related to data transport. Such services are, for example, modifying network data, signaling, signaling error correction, etc.

As a result to cooperation and composition of these services, they define workflows, which provide more complex functionalities by selecting sets of services and defining their interaction. Thanks to the loose coupled definition of services and the abstraction of their functionalities SONATE tries to simplify the service composition and orchestration, permitting to add or remove services and change their implementation without affecting the whole workflow.

Selection and composition are the tasks of finding a building block graph to fulfill a certain need. A workflow can be defined explicitly by a requester or may be derived from known dependencies. Automatically creating a workflow on demand, e.g. connection setup, offers the highest degree of flexibility and workflow optimization, responding in short-term to user requests or changing environmental conditions.

Any algorithm that tries to solve the composition problem needs information about which service connections and graphs are valid and which attributes that combination has. This information must be provided in a way that can be processed automatically and it must not restrict service interaction in any way. The easiest approach would be to list all services that are compatible together with the attributes of the combination. For some usage domains this is feasible but not for a future network architecture. A cornerstone point that SONATE tries to enhance is the flexibility of the composition (and thus the network). In this sense, they consider that a global database where all dependencies and combination possibilities of services are to be stored would destroy this flexibility. This means that dependencies and combination possibilities must be expressed in an implicit way that does not require any changes to other services when one service is added, changed or removed.

II.1.2 Service Composition Representation

For building a network based on SOA principles they consider specific supporting techniques, since Web Services and XML are inappropriate to implement services on a network level (other light-weight semantics are needed). However, for primary implementations of their framework, they specify services and workflows using XML.

II.1.3 Service Composition Process

For each set of requirements of an application, a matching set of services/mechanisms could be assembled. Those mechanisms then form a custom micro-protocol stack (which is more like a mesh in fact).

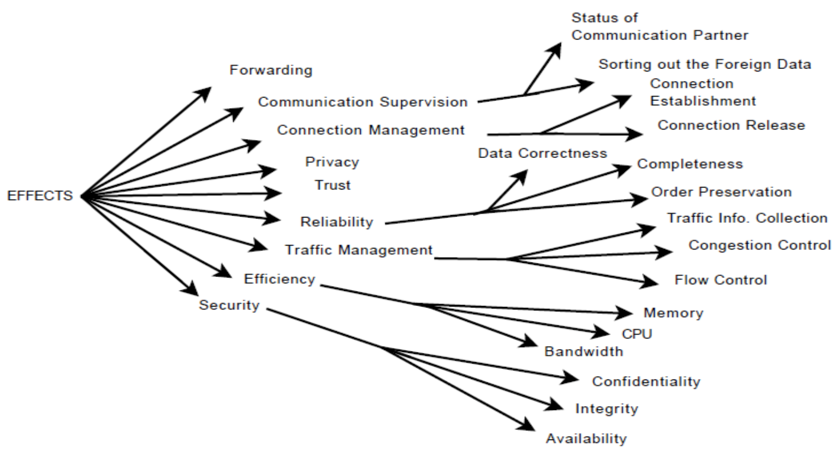


Figure A.6 Effects defined in SONATE [111]

To dynamically build these micro-protocol stacks SONATE needs to describe their capabilities and requirements in a way that an algorithm can predict the outcome of any combination. To achieve they specify that each mechanism has a set of requirements, a set of provided effects (Figure A.6), a description of the costs of the mechanism and a formula how these costs "add up" with the costs of other mechanisms. An algorithm selects mechanisms so that all requirements are met, all desired effects are provided and the combined costs are low.

One approach to do composition process is presented in [111]. It bases the composition process on a network architecture that is described as a 5-tuple (*E, M, M_f, P, R*). *E* and *M* are sets of all available effects and mechanisms. For instance, *E* could contain *Completeness* and *Order Preservation* and *M* contains *Sequence Numbers*, *Acknowledgments* and *Header Checksums*. $M_f \subseteq M$ is the set of mechanisms that represent lower network layers and therefore it is fixed. *P* is a function $P: M \rightarrow 2^E$ that gives for each mechanism a set of effects that it provides (e.g. *Sequence Numbers* provides *Order Preservation* and *Acknowledgements* provides *Order Preservation* and *Completeness*). And finally, *R* is a function $R: M \rightarrow 2^M$ that gives for each mechanism a set of other mechanisms that it requires. For simplicity, the functions: $P^*(M \subseteq \mathcal{M}) = \bigcup_{(m \in M)} P(m)$ and $R^*(M \subseteq \mathcal{M}) = \bigcup_{(m \in M)} R(m)$ are defined.

II.1.4 Service Selection

A given set of mechanisms $M \subseteq M$ is consistent if all requirements are met:

$$consistent(M \subseteq \mathcal{M}) = R^*(M) \subseteq P^*(M) \cup M$$

To compare different consistent sets of mechanisms a cost function is used. *C* (cost) is a function $C: 2^M \rightarrow R$ that values the combined costs of a set of mechanisms. Since different types of costs “add up” differently (e.g. latencies sum up but loss ratios are inverse multiplicative) and the factors with which these types of costs are combined depend on the needs of the application layer.

To improve this model, *meta-effects* (effects entirely composed of other effects and meta-effects) are introduced. E_M is the set of meta-effects and *ComposedOf* is a function *ComposedOf*: $E_M \rightarrow 2^{E \cup E_M}$ that gives for each meta-effect a set of effects and other meta-effects that it is composed of.

$$ComposedOf^*(X \subseteq \mathcal{E}_M) = \bigcup_{x \in X} ComposedOf(x)$$

Moreover, for simplicity reasons it is defined:

$$ComposedOf'(X \subseteq \mathcal{E}_M) = ComposedOf^*(X) \cup \mathcal{E} \cap ComposedOf'(ComposedOf^*(X) \cup \mathcal{E}_M)$$

This is the transitive hull of *ComposedOf* that only contains effects as result. With this extension to the model *meta-effects* like *Reliability* can be constructed with the effects *Data Correctness*, *Completeness*, *Order Preservation*.

10.1.1.1 Service Selection Process

Given a cost-function *Cost* and a set of desired effects *D* the function:

$$S(D, Cost) = \arg\min_{M \subseteq \mathcal{A}} Cost(M)$$

With,

$$\mathcal{A} = \{M \subseteq \mathcal{M} | \mathcal{M}_f \subseteq M \wedge consistent(M) \wedge D \subseteq P^*(M)\}$$

This returns the best set of mechanisms to satisfy the requirements. Calculating the result of $S(D,Cost)$ is a task of optimization. In SONATE project, as far as the authors know, it is only presented a simple approach:

1. Select mechanisms to provide the requested effects. The base technology already provides some mechanisms M_f. This will yield a number of different premature solutions. At this step the premature solutions are not checked for consistency, they provide the requested effects, but their requirements might be missing.
2. Create consistent solutions and rate them. Mechanisms are added to the premature solutions to make them consistent. After this step a number of consistent solutions is known and compared using the cost function.
3. Improve solutions with additional mechanisms. In this final step the best solutions are improved by adding mechanisms to them.

The goal is to improve the cost of the solution without losing the consistency. Another possible solution is to have a pre-calculated list of solutions and to adapt them in a predefined way, as the Netlets in 4WARD project. Or maybe multiple heuristics can calculate solutions concurrently and then the best one is chosen.

II.2 4WARD

One of the Future Internet visions of 4WARD (it is a large European project that does not define a unique approach for Future Internet) specifies a framework very similar to G-Lab project. Mainly, the composition of functionality is based on chaining some Building Blocks or Functional Blocks which are implemented by specific Functional Block Mechanisms. 4WARD is also based on a Component Based Architecture (CBA), where the decomposition of the engineered systems into functional or logical blocks with well-defined interfaces used for communication across these components. Components are considered to be a higher level abstraction than objects and as such they do not share state and communicate by exchanging messages carrying data.

II.2.1 Composition framework

In 4WARD the dynamic behavior is provided by a set of policies specified by each Building Block (BB) which are evaluated at run-time. An example of their appliance would be when it is necessary to choose a polynomial to calculate the CRC in the presence of an alarm or specific event.

4WARD organizes Building Blocks in different strata (Horizontal, Vertical and Abstract). In addition, 4WARD is able to compose context-aware services thanks to the concatenation and execution of BBs which perform a specific action is contained and specified in a Netlet. 4WARD proposes to use a repository of well-known or best practices to effectively compose different types of functionalities, it is called Design Repository.

In order to cope with the high complexity of communication systems, 4WARD provides two different views on network architectures: the macroscopic view and the microscopic view. The first one is more related to an overall structuring of the network architecture at a rather high level of abstraction in terms of strata. The second one deals with the functionalities needed within network architectures, their composition to so-called Netlets in order to fulfill desired requirements as well as the Node Architecture hosting various Netlets of the same or different families of network architectures. The microscopic point of view introduced in 4WARD tries to tackle a close problem of this research, composing basic functionalities or atomic services into more complex services. The goal of the composition is to build new protocols fulfilling the requirements of each communication.

II.2.2 Composition representation

In 4WARD, functionality is represented as shown in Figure A.7. The functionalities used in a network can be part of different categories. Each category can contain different functional blocks and a functional block can belong to different categories at the same time (Figure A.8).

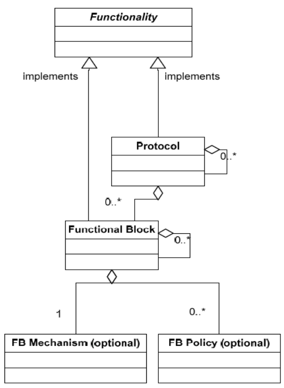


Figure A.7 Functionality decomposition

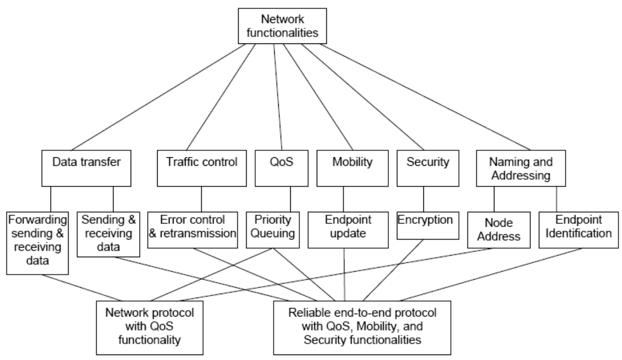


Figure A.8 Functionalities composition

The ordering of functionalities is important, it must consider hop-by-hop communications and end-to-end communications restrictions.

II.2.3 Service Composition Process

To select the best functional block among all the possible ones, a model for representing them is needed. 4WARD model tries to describe direct and indirect changes that a Building Block does to data, thus, they are represented as black boxes (Figure A.9).

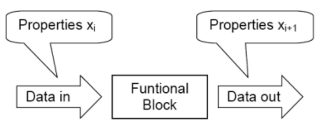


Figure A.9 Functional Block abstract model in 4WARD

The model of the functional blocks is used as a base towards selection and configuration of functionalities, which is based on [112].

By calculating what each functional block does to the data, a prediction of what a composition does can be obtained. This prediction is precise enough to compare it with the predictions of alternative compositions, therefore allowing us to choose the "best" composition for a given data flow.

Regarding to protocols, they are basically means of data exchange for a communication between nodes or systems. In order to understand each other, the concerned communication parties must agree on syntax and semantic of exchanged data units. Syntax means the data unit's structure and the fields it contains. Semantic means the meaning of each field of the data unit. Syntax and semantic make one protocol different from another and create a variety of network protocols. Current data units are usually composed of payload and header. The header in turn is composed by fields carrying control information such as network address, data checksum or congestion notification. Based on the header structure presented in Role-based Architecture (RBA), where roles can be equivalent to functionalities in the context of this section, they propose a method to design new protocol and dynamically build the protocol header by functionality composition. A protocol implements a set of abstract functionalities and is composed by one or many FBs. The header is built based on the functionalities needed by composing the corresponding functional blocks, and finally associated with the corresponding payload to form the whole PDU of the protocol. The separation between the functionality and the functional block can provide the protocol design and header construction with a high flexibility similar to the concept of polymorphism that exists in software engineering. As an example, many addressing schemes may be used to implement the addressing functionality and each provides an address value, which can be transported by the header. When switching the address scheme, the address format and value in the header change accordingly. One could achieve this by implementing the addressing scheme as Functional Block Policy.

Regarding the communication related functionalities; different FBs can implement a needed functionality. The protocol specification and the header format will change accordingly to the functionality composition in order to get the required protocol (Figure A.10).

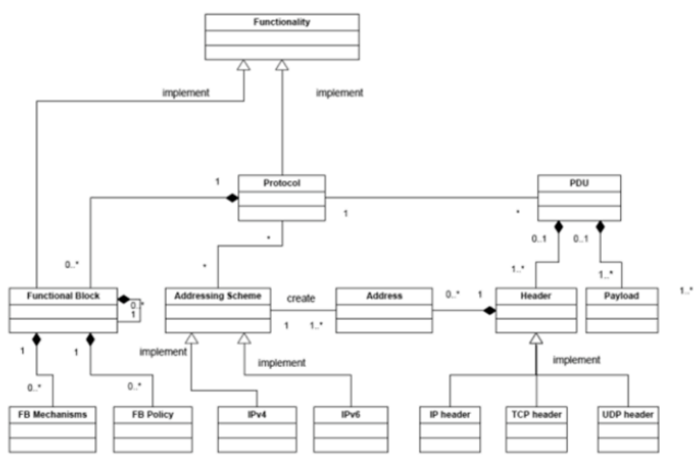


Figure A.10 Functionality abstract model in 4WARD

II.2.3.1 Service Selection Process

The selection algorithm of the node architecture is based on the selection algorithm of Auto-Configuration for Communication Security (ACCS), which they have designed to evaluate the effects and side-effects of adding security protocols to a TCP/IP stack. The selection algorithm compares different candidates and chooses the optimal one, based on user or application requirements. A candidate in the ACCS context is the combination of TCP/IP and one or more security protocols as well as a configuration of that protocol. For the node architecture these candidates are more flexible, they are Netlets that are possibly composed of functional blocks.

When trying to find the optimum, it is important to find a trade-off between different properties of the candidates. Special care is needed for security, therefore, they have introduced a special security property in [112] called Effective Bit Strength (EBS), which will be also used for the new solution. The main idea behind EBS is that an encryption algorithm, for example, is only as strong as the easiest attack against it, even if that means attacking the key exchange.

In the first step of the evaluation of candidates, they filter all candidates based on system policy or user/application requirements. This means candidates that cannot fit the requirements will not be considered later on. The requirements are basically restrictions on the properties of the Netlet, e.g., the energy consumption must be below 100 J.

Based on these properties, they use a well-known decision-theoretic method, e.g., multi-attribute utility analysis (MAUT) [113], to aggregate the results of a candidate with respect to each single property into an overall ranking of each candidate. MAUT structures a complex decision process, which depends on multiple attributes, into a per-attribute utility evaluation. Therefore, for each attribute a utility function has to be defined which maps the value of an attribute to its corresponding utility. A simple exemplary utility function is shown in below. For valid attribute values, e.g., latency values smaller than the defined maximum, the utility increases with decreasing latency. The curve models the fact that smaller latency values are linearly preferred over bigger latency values. Subsequently, to get an overall ranking of the available candidates, the utility values of a candidate's attributes are aggregated using the weighted sum as aggregation function. By adjusting the weights, one determines the relative importance of attributes, e.g., the share they contribute to the final utility of a candidate. For n attributes a_i, n utility functions $u_i()$, and n weight parameters α_i where $1 \leq i \leq n$, the overall utility U of a candidate is then expressed by:

$$U = \sum_{i=1}^{n} \alpha_i * u_i(a_i)$$

This overall rating is then calculated for each candidate and the highest-rated candidate is then chosen for the communication association (Figure A.11). Additionally, in order to determine the overall behavior of a Netlet, 4WARD defines the following notation (Figure A.12).

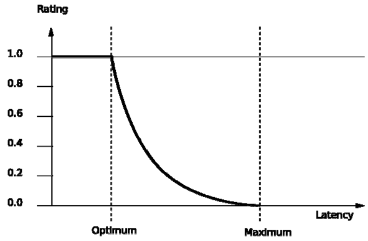


Figure A.11 Rating function

β_i	Functional Block i
Ψ_i	Effects of Functional Block i
Ξ	Global State
$\vec{\rho}$	Properties of the data stream
$\vec{\rho}_i$	Properties after Functional Block i
f	Transfer function

Figure A.12 Functional block parameters

This algorithm was not only designed to allow the node architecture's decision engine to choose between different Netlets at runtime, but it was also designed for performance.

Without loss of generality, assume that functional blocks β_i are ordered corresponding to their order of execution. The input to Netlet 1 is then processed by the functional blocks 1 to n, which affect the input while passing through with Ψ_1 to Ψ_n, respectively. 4WARD expresses the experienced effects of a functional block to the properties ρ using the transfer function shown as: $\vec{\rho}_i = f(\vec{\rho}_{i-1}, \Psi_i, \Xi)$

The transfer function is parameterized using Ψ_i, which describes the effect of a functional block on the properties. The global state contains information about the node and its interfaces. The overall effects of a Netlet on the properties can then be described by: $\vec{\rho}_n = f(f(...(f(\vec{\rho}_0, \Psi_1, \Xi), ...), \Psi_{n-1}, \Xi), \Psi_n, \Xi)$

where ρ corresponds to initial properties of the input. These initial properties also include a traffic profile of the application, e.g., a discrete packet size distribution.

Next some examples are shown in order to better understand the determination of the properties of a Netlet. The first example will focus on latency of the data processing. Since latency is cumulative, the latency of a Netlet can be estimated by summing up the Functional Blocks' individual latencies. So the effect on the property would be based on the add operation, as in: $\rho_n^{la} = \rho_{n-1}^{la} + \Delta_n^{la}$

Since most applications do not only communicate with a single data sizes, they use a traffic profile. The traffic profile specifies the distribution of a limited number of data sizes, representing the expected traffic. Bulk data transfers, for example, try to transport a large amount of data in one direction and only transfer a small amount of control data in the opposite direction. When looking at the data units, it is clear that for one direction the probability of large data units is high and for smaller ones low, while for the opposite direction the probabilities are the opposite: a high probability for small data units and a small probability for large data units.

Another functional block might not just increase the length by a constant Δ_i but could increase the data length based on the original data length. Block-based cryptography, for instance, might need to pad the data before processing; effectively increasing the data to the nearest block size b:

$$\rho_i^l = \lceil \frac{\rho_{i-1}^l}{b} \rceil * b$$

Also a decrease in length is possible, for instance by using data compression. However, one must differentiate between two cases: predictable compression and unpredictable compression. In the first case they can predict the changes to the length. This is usually the case with header compression algorithms. The unpredictable compression, however, is more difficult to handle. Some data compression algorithms could even increase the length by adding meta data, when the data is uncompressible. While this can be modeled with stochastic models, the negative effects of the unpredictability of data compression may be reduced by allowing the data compression only to be used on data streams and not on data units. Negative runtime effects of the increased data unit size, like today's fragmentation, can be also avoided this way.

The previous examples showed how the functional building blocks influence transported data. Energy Consumption and Length are just examples for influenced properties to present our approach. 4WARD has also looked at other properties, such as latency jitter, bandwidth, packet loss rate, bit error rate, and effective bit strength.

II.3 Service Composition Conclusions

Instance and template based approaches are not suitable for the solution we are trying to adopt as they do not provide enough dynamicity and flexibility, demanded as a strong requirement for future networks

Declarative approaches can offer near-to-optimum compositions when having a cost function. For example, FUSION [84], although it is focused on web services, it introduces a framework very similar to what is foreseen to be used for networking services. It works with a close bundle of services that are previously defined and stored. This knowledge allows delimiting the number of compositions. Moreover, an important feature of this composition system is that it is able to verify if the result of the execution plan that generates meets the requirements of users; and if not, it immediately performs again another composition.

However, it should be interesting to add additional features to empower the user selection, by providing a more accurate description of context and services. In that sense, the revision of ontology-based approaches gives guidelines on how to declare. For example, DAML-S and its successor, OWL-S, present very complete ontologies for services, and a classification of them in atomic, simple and composite services, but they are based on Web Services, and it would be inefficient to map all network service parameters in them due to the heavy format they introduce. Regarding to context parameters, and taking into account all the groups defined by Dey in [104], the ontologies defined by WSOL should support all of them. At this point, it should require a more extensive analysis once services and context parameters are delimited. As alternative, service and context ontologies could be defined taking into account only the parameters needed using logical language as done in [64]. This representation can also include order restrictions in services, permitting to automate the composition process only considering knowledge parameters from service and context.

Regarding to AI Planning and other hybrid techniques, they should be considered for optimizing composition processes in future steps but most of them are out of scope at this point.

Additionally, the two revised Future Internet projects present similar approaches, in the sense that both use a similar set of services (mainly based on Role-Base Architectures). However, SONATE tries to implement a more dynamic service composition process defining a set of effects to have a close control of the iterative composition process, while 4WARD performs it in a more statically way, not permitting a real automatic and dynamic composition at run-time. For service selection, both projects use the maximization of a cost function, but 4WARD tries to take into account more user and network requirements obtained from context parameters.

APPENDIX III List of Atomic Services

This appendix makes an overview of some services defined in different proposals.

In the Deliverable TWP1.1.D3 Atomic Service specification (TARIFA Project, i2CAT Foundation) authors list the following services:

1. Encoding and signaling

   This service is responsible for various encoding and signaling functions that transform the data from bits that reside within a computer or other device into signals that can be sent over the network.

2. Data transmission and reception

   This service is responsible of actually transmitting and receiving encoded data through the physical channel (wired or wireless).

3. Data Framing/Encapsulation and Unframing/Decapsulation

   This service is responsible for encapsulating or framing data streams into discernible blocks of information or messages.

4. Error Detection and Handling

   This service is responsible for detecting and handling errors occurred during message transmission.

5. Medium (Multiple)Access Control

   This service is responsible of managing the procedures used by devices to control access to the network medium.

6. Sequencing

   This service is responsible of tagging chunks on transmission with sequence numbers.

7. Message Retransmission

   This service is responsible of controlling message/chunk retransmission at other atomic services request (like ARQ, flow/congestion control, etc.).

8. Acknowledgements

This service is responsible of acknowledging received chunks in order to guarantee reliable delivery of data.

9. Flow Control

   This service is responsible of controlling the sender's traffic rate in order to allow the receiver to handle all the received data.

10. Data Forwarding

    This service is responsible of forwarding data messages to its next destination.

11. Fragmentation and Reassembly

    This service is responsible of fragmentation data units (chunks and objects) into small chunks in order to fit transmission requirement of maximum payload size of underlying carriers (frames in case of chunks and chunks in case of objects).

12. Data Demultiplexing and Multiplexing

    This service compresses the combining of two or more information channels onto a common transmission medium and then recovering the separate information channels at the receiving end.

13. Addressing

    This service compresses assigning logical addresses to network entities for its location and/or identification in network.

14. Connection establishment, management and termination

    This service deals with the establishment of connection and its management and termination in connection-oriented communications.

15. Congestion control

    This service is responsible of control network congestion when it happens or preventing network congestion before it happens

16. Access Control

    This atomic service is responsible of controlling access to consumers requesting to consume a certain composed service.

17. Data Encoding and Decoding

This service is responsible of encoding data with suitable encoding scheme, transforming raw input data into desired format.

18. Data transcoding and Translation

    This service is a chain of decoding/encoding services with the purpose of transform data from one format to another different format

19. Semantic Identification

    This service is responsible of resolving semantic identification requests during session negotiation

20. Caching

    This service is responsible of caching transferred objects through a session, storing a copy of the object for later use.

21. Object flow

    Object flow service is responsible of manage data objects transmission in order to transfer data objects as a stream of objects.

APPENDIX IV Technical aspects of service composition

The process of combining available Atomic Services to create a desired communication service is called service composition.

As opposed to the stringent protocol-oriented approach of current TCP-/IP based communications, the proposed service-oriented and functionality-composed approach adopts a loosely-coupled design. As such, it is beneficial in many aspects:

- It is flexible in building multi-feature and customized communication services
- It allows users to participate in the provision of the services they desire (e.g. user-control routing for performance or cost reasons)
- It provides for adaptation to heterogeneous networks from the very same terminal device
- It facilitates the deployment of new network, service and/or information access technologies from network and user access perspectives
- It avoids redundancy of functionality both in terms of duplication and unnecessary placement

The FN service composition prompts for baring user devices and smart networks. Smart networks would equip on-the-fly with the necessary communication functionalities as appropriate to changing user needs and requirements while, at the same time, they would choreographise themselves to deliver the requested services at the desired quality levels.

Evidently, due to performance and scalability reasons, service composition cannot be feasibly implemented across all communication layers. Performance-critical communication services need to cast to hardware or firmware e.g. those related to transmission; as such, they are better shipped with the hardware interfaces (NICs) than being composed. Services at the network or, even, higher communication layers, being implemented in software, can well be composed rather shipped in monolithic stack-based systems. Thus, to accelerate composition time, predefined service compositions can be stored (e.g. in templates) to be reused avoiding calculating them again.

The following sections outline the technical issues pertinent to communications service composition.

IV.1 Composing network functionality

A service composition process is always driven by a target/desired communication service. For instance, when a user requests a service or an in-network QoS-aware connectivity service that needs to be provided under certain conditions and constraints.

The target/desired communication service must have specific and well-defined objectives. Its conditions reflect the situational state of the surrounding environment at the time of its usage, which can be modelled through context information. And, its constraints reflect user preferences and network technological capabilities. All this information constitutes the input to the composition process.
No matter where and when executed, a composition process implies the following steps:

- Capture of input information; target service objectives, context information and network technology capabilities.
- Identification of the service functionalities to making up the target service based on information from corresponding to the desired service type communication templates and information from matching the required with the offered service properties.
- Discovery the identified service functionalities, within and across provider domains.
- Selection of the most favourable synthesis using the discovered service functionalities, on the basis of current component availability and operational efficiency metrics; in the general case, there may be alternative synthesis patterns and/or alternative service functionalities per synthesis.
- Allocation and realization of the determined synthesis by enforcing the interoperation of the selected service functionalities.

FN architecture will specify and develop suitable means for effectively and efficiently realize the functionality involved in the above composition functions.

IV.2 Composition scope and service granularity

The FN is concerned with the composition of the required functionality to accomplish user communications, out of a set of existing basic data-networking-level service functionalities e.g. data transfer, forwarding or naming. Hence, the FN focuses on composing services at the connectivity, transport and application communication levels.

Independent of network or service technology, the FN considers basic service functionalities that can be classified as in two levels:

- network (connectivity services)
- higher communications levels (transport and application services)

Furthermore, services can be part of two spaces, user device and network spaces. The term in-network services to refer to services/functionalities that reside in the network. An initial view of assumed existing service components is depicted in the following Figure A.13.

The FN will be agnostic to the underlying technology of the existing service functionalities to be used in composition, while it does not place any requirements or constraints on the functional granularity of these services. The FN is not focused on standardising the set of basic networking service functionalities, that is their functional granularity. FN is concerned on the development of generic means to intelligently bundle together sets of available service components to eneable end-to-end communications.

There should be a trade-off associated with the functional granularity of the service functionalities to be synthesized. The finer the granularity, the more flexible the composition outcome, but the more complicated and time consuming the composition process would be. The coarser the granularity, the less flexible the composition outcome, but the more simplified the composition process is.

IV.2.1 Place for composition

For a particular target communication service, such as a required end-to-end user service like a streaming service, composition processes need to take place both at user and network spaces.

At the **user space**, the composition process is executed at end devices, based on services residing therein and on in-network services that need to be shipped by the network to the user devices. Alternatively, the network or networks where the user is connected to may compose for the user the required communication services and ship the resulted service to the user device. Evidently, composition at the user space enables users to have control over their communications. For instance, this is specialy interesting in multihomed scenarios.

In any case, by relying on composition, user devices should not commit to any communications stack, above a basic networking level. Instead, they should be equipped with the necessary functional and execution environment to enable communication service composition or undertake composed services from the network.

At **network space**, the composition process may be executed centrally or in a distributed fashion within a provider domain, depending on the nature of the communication service to compose and the engineering strategy and architecture of the provider. Composition at different provider domains may also be required, for instance, achieve an end-to-end QoS-aware connectivity service requested by a user.

Once a service is composed, as decided by the composition process, it is realized in a distributed fashion, spanning from user devices to end service nodes across different provider domains, using the constituent services along the way.

The realization of a composite service and ist management are important aspects of the composition process, as they affect the performance of the resulted service. Deciding which constituent services to use in the composition process should take into account not only functional relevance but metrics related to their realization (e.g. set-up time) and its operational efficiency (e.g. robustness, energy consumption, etc.).

IV.2.2 Composition execution epochs

There should be a trade-off associated with the time required to execute the composition process with respect to the time a requested service is called. Basically, there are two extreme ways to do service composition.

- **Design time execution**: the composition process has little information to utilize, therefore its outcome might not meet user expectations, but it has sufficient time to generate a possible solution.
- **Run-time execution:** the composition process has more information to utilize, therefore its outcome may be more accurate, but it has to be realized in virtually zero time.

Run-time execution might be suitable, within narrow time margins, for explicitly called services which, involve a signalling phase (e.g. voice services). However, it is totally unsuitable for most Internet information access services, where data is transferred in a connection-less mode.

Moreover, the FN architecture would be able to rely on hybrid schemes combining the merits of both approaches desing-time and run-time composition execution schemes. Hybrid schemes can be based on a 'prepare-to-fly' approach, versus a truly 'on-the-fly' composition approach, where compositions can be prepared in advance to eliminate additional composition overhead.

To achieve this, a possibility would be to adopt a gradual composition logic evolving over different time epochs.

At the **user space**, composition may take place gradually at the first time the user end device is connected to the network and at current service demand time, or, even, the first time a service is requested.

At the network space, in-network service composition may take place at various epochs so that to create an adequate readiness level. Appropriate in-network services may be composed at design times to have efficient and reliable realization times.

To practically implement this evolutionary composition approach, we introduce the notion of communication templates. A communication template prescribes a specific set of atomic services to be used in the composition of a particular type of desired service. The constituent services will be completely specified during the composition process, where the current details and characteristics of the requested communication service will be known (e.g. context parameters). These templates are created manually at design time, reflecting the knowledge of the network provider to provide a certain set of service types, based on technologies that employs or can support. In addition to service types, communication templates may be dependent on the epoch to perform composition, prescribing more gross guidelines. Suitable means to describe and efficiently interpret such guiding templates for optimizing the performance of the composition process will be needed.

FN must provide a complete a framework including network architecture, protocols, and corresponding processes. This framework will stress on the network composition, permitting to compose services in three levels: connectivity, transport and application. With this in mind, FN architecture present both, evolutionary and clean-slate approaches. It can go for evolutionary or incremental approach, composing transport services and assuming that network technology is in place. However, it could be seen as a revolutionary architecture when connectivity services are composed as well.

A three-plane architectural view can be adopted. At the bottom, *a data plane,* distributed throughout network nodes, is responsible for transferring the data of user services. This plane is built based on existing link transmission and switching technologies. On top of it, a *service control plane*, at each network node, runs the logic of the distributed service composition protocols and appropriately configures the data plane for constructing the paths along which the end-service data will flow. The protocol messages are transported from node to node by means of the data plane and their routing is determined by the logic of the protocol. Finally, at the top, a *context plane*, residing in network nodes and/or in an overlay, maintains the information required by the protocol logic to route its messages; the plane is also updated with service-related information from the service control plane.

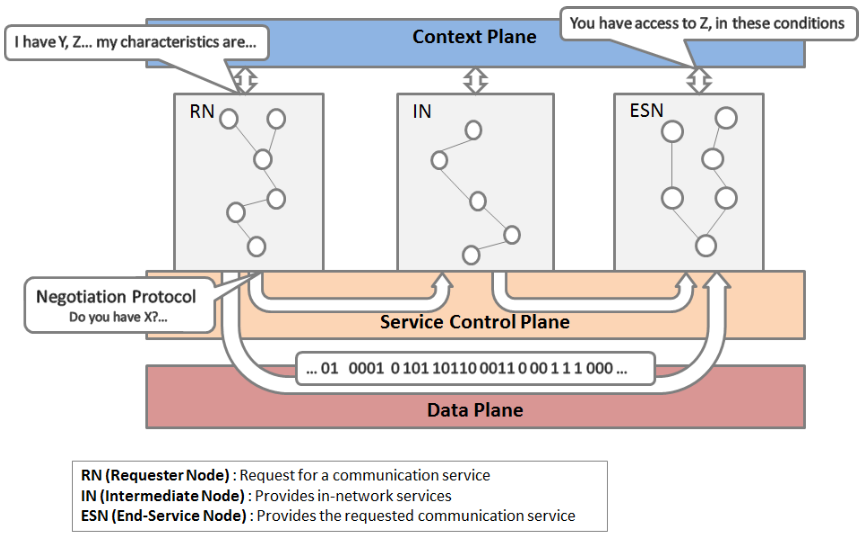


Figure A.13 FN architecture communication planes

FN will need to explicitly address composition in a multi-provider environment, seeing into the necessary interaction between providers and allowing orchestrating services that involve entities of different domains and providers to provide end-to-end services. Some entities will be defined during the project for assuring this functionality. However, an envisioned view of a scenario can be seen in Figure A.14, where manager entities of the domain will share some information such as available services of the domain with associated costs and QoS provided. To achieve this, appropiate SLA agreements must be arranged between the involved and inter-connected entities.

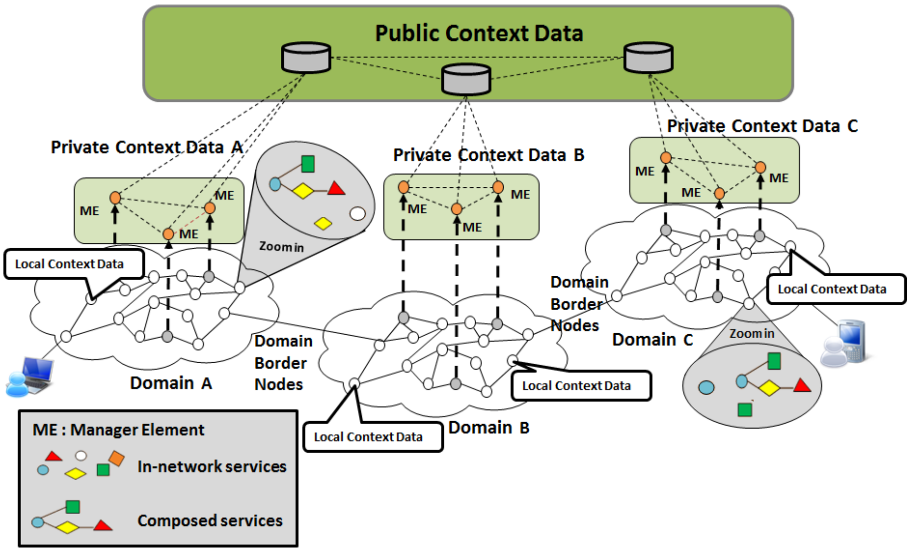


Figure A.14 FN architecture in an infrastructure multiprovider scenario

In order to operate with such scenario, two kinds of protocols will be needed:

- Distributed service composition protocols for composing services at the connectivity level; the resulting composed service determines a path for specific service data flows (the protocol will be context-aware integrating semantic service discovery capabilities.)
- Negotiation protocols that permit providers to negotiate configuration parameters of the in-network services and establish SLAs between them.

IV.3 Composition of transport and application services

The FN architecture will have the ability to reconfigure the network for each demanded service and adapt network characteristics to each separate communication flow.

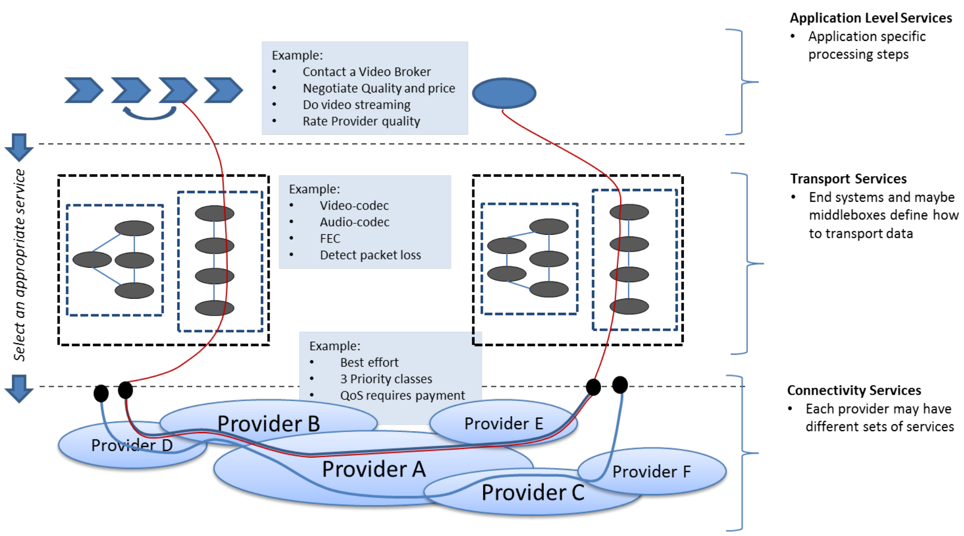


Figure A.15 Services at application, transport and connectivity levels

FN service composition considers three levels of network services (see Figure A.15), whereby each level represents the domain of different service stakeholders:

- **Connectivity services** represent the services of network infrastructure providers, e.g. ISPs. Examples of connectivity services are: best-effort end-to-end delivery, per flow QoS or use of traffic classes. In addition also services of new architectures like Information Centric Networks (ICN) are considered.

- **Transport services** represent features of end-to-end data transport as used by applications. Transport services utilize connectivity services and usually add functionality within the hosts and maybe middle boxes. Examples of transport services are reliable data streams (as provided by TCP) as well as delay tolerant packet delivery or bulk data transfer optimized for high-speed networks.

- **Application services** are related to specific application tasks and may make use of services residing within a network. Application services utilize transport services and several application services can be composed to a new more complex application service. An example of a composed application service is the whole process of a video on-demand process, including discovery of a video provider, billing and video transcoding.

In order to provide an optimal service to end users, service composition and service adaption can take place on all three levels and each level can select an appropriate service of the next lower level. FN architecture follows an integrated approach by considering all levels of services. The key to achieve this is a clear and common definition of network service descriptions. Suitable service descriptions must be provided by the FN.

Optimal service composition requires solving the trade-off between: flexibility, temporal overhead and performance. On the one hand, late composition (like late-binding) enables highest degree of flexibility by taking into account more and more accurate input data. On the other hand, at run-time service composition must be performed in few milliseconds, rarely a few seconds are acceptable. FN architecture will could facilitate this trade-off by splitting the service composition into several epochs (e.g. design time, deployment time, run time, etc.). Complex and long running tasks will be performed "early" while simple and thus fast tasks can be performed "late". Note that service composition also involves the task of selecting a service of the next lower level (Figure A.16).

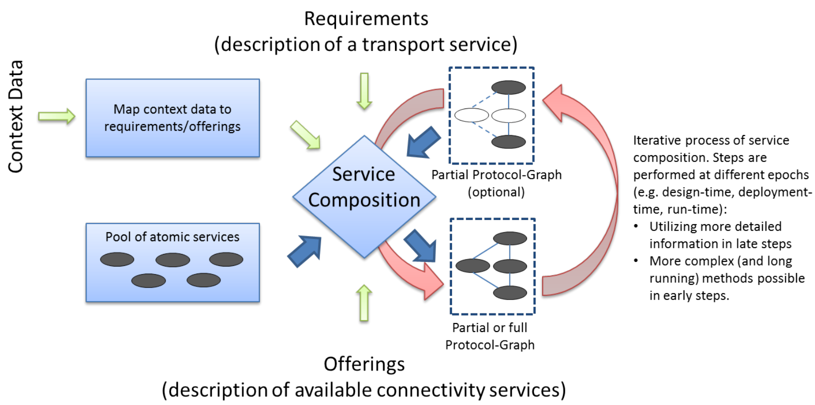


Figure A.16 Service composition process overview

Each service level requires a different service composition approach because: Connectivity services are made up of existing infrastructures and thus cannot change functionality on application demand; Transport services are composed of functionality (e.g. protocols) provided by involved hosts; Application level service may spawn a very wide range of service types and make use of arbitrary services provided in the network.

The composition process is driven especially by application requirements and context data and, of course, by the availability of appropriate atomic services.
Network context will be considered taking into account characteristics such as network topology, links' bandwidth, physical layer characteristics and available resources of demanded networks and devices.

Networks could have live information of the network state and be always aware of resources with no added overhead or very little overhead.

Moreover, FN architecture will provide mechanisms allowing applications to notify their requirements and preferences to the network. It is necessary to introduce new expressive formats, including semantics and ontologies, that allow users/applications to say to the network what it wants and under which conditions. These requirements will be evaluated during the discovery phase in order to find the best nodes and, consequently, the paths until the end node providing the demanded end service.

However, in order to enhance user decisions but to limit the information transferred in the network a set of policy rules will be defined and imposed to not collapse networks with an overload of data to be processed.

APPENDIX V Adaptive Beaconing Algorithm

This appendix makes an overview of the Adaptive Becaoning Algorithm published in [GP5]. For further details and results, see [GP5].

V.1.1 Introduction

The number of vehicles contending for space in existing transportation systems is growing rapidly. This abrupt growth of vehicles has made driving unsafe and hazardous. Thus, existing transportation infrastructure requires improvements in traffic safety and efficiency. To achieve this requirement, Intelligent Transportation Systems (ITS) have been considered to enable diverse traffic applications such as traffic safety, cooperative traffic monitoring and control of traffic flow. These traffic applications can become realities through emerging VANET because vehicular network is considered as a network environment of ITS. In addition, in the near future more vehicles will be embedded with wireless communication devices such as Wireless Access in Vehicular Environment (WAVE). When vehicles are equipped with WAVE, they can synchronize and handshake via beacons. In this way, a vehicle exchanges beacon messages periodically, sharing its mobility characteristics with its neighbours, thereby building cooperative awareness. However, rapid changes in traffic density from sparse to heavy, as well as periodic beaconing between vehicles can cause the wireless channel between vehicles to promptly become congested, resulting in a high degree of performance degradation of vehicular network. The reason for this channel congestion is that each vehicle periodically broadcasts beacons at a fixed rate. This also leads to high channel overloading and hence packet loss. In short, the higher the frequency of beacon rate, the higher the bandwidth overload in dense traffic conditions. On the other hand, the solution to channel overloading does not involve simply reducing the frequency of beacon generation. As the frequency of beacon generation is reduced, the error will increase between the current physical position and the last reported position. For instance, in geographical routing protocols, reducing beacon rate would lead to the inaccuracy of the exchanged position coordinates between vehicles. This would negatively affect the performance of routing protocols. In short, reducing the beacon rate leads to the exchange of out-of-date information. From the brief discussion above, it is obvious that there is a pressing need to consider a conditional update approach in which a vehicle adapts its beacon rate when there is considerable variation in its neighbour vehicles mobility/traffic characteristics. Therefore, multiple parameters, like vehicular mobility characteristics and status of vehicle, have been utilized to design an intelligent ABR approach to control beaconing rate. This is because a fixed beacon rate can not tackle both bandwidth consumption and accuracy of vehicle status due to rapid changes in vehicular traffic conditions. Therefore, an intelligent ABR approach in vehicle-to vehicle communication has been developed to tune the beaconing rate in response to changing vehicular traffic characteristics. The contributions of this study can be summarized as follows:

1. In dense traffic conditions, a low beacon rate is required to reduce overload on the network (with acceptable information awareness) whereas in sparse traffic conditions, a higher beacon rate is required to increase the cooperative awareness (with acceptable beaconing load) between vehicles. Therefore, in contrast to all previous works, we proposed an intelligent ABR approach based on fuzzy logic to tackle the aforementioned issues.

2. We perform simulations to show the effect of traffic density, number of emergency vehicles and shadowing lossy channel on the proposed approach. In addition, the proposed adaptive approach has been modeled and simulated using JIST/SWANs simulation tool for performance evaluation. Likewise, the fuzzy logic decision-making algorithm—which is integrated with the ABR approach—is implemented in java language.

V.1.2 Proposed Adaptive Approach

The designed ABR (Adaptive Beaconing Rate) approach is adopted for Vehicle to Vehicle (V2V) communication systems in which vehicles communicate without the presence of infrastructure. The approach is used to tune the frequency of beacon generation with traffic context in VANET. We assume that all vehicles are equipped with wireless radio communication devices in order to facilitate communication with other vehicles. Similar to existing work on VANET, we assume that all vehicles are equipped with a Global Positioning System (GPS) receiver that provides vehicle position information. We also assume that different types of vehicles are deployed in the urban area to account for the presence of both emergency and non-emergency vehicles. Since vehicles on the roads are susceptible to unusual situations, the presence of emergency vehicles is a reasonable assumption.

Instead of simply broadcasting beacons in a fixed time interval, we propose a VANET friendly adaptive approach to control beacon rate. Whenever a vehicle receives a beacon message from its neighbours, the vehicle checks the percentage of directional neighbour vehicles and its emergency status. After collecting this information, it triggers the fuzzy inference system (it is run distributedly by every node upon receiving a periodic beacon message) to calculate the value of the required Beacon Rate *(BRr)*. The new value of Beacon Rate *(BRn)* is then calculated based on the following equation:

$$BR_n = BR_c + \gamma(BR_r - BR_c)$$

Where *BRn* is the new value of beacon rate, *BRc* is the current value of beacon rate, *BRr* is the required beacon rate which is the output of fuzzy inference system. Further, γ is the weight factor which is used to sustain the value of *BRn*. If the value of $\gamma = 0$, *BRn* = *BRc* i.e. it negates the effect of beacon rate adaptation. On the other hand, $\gamma = 1$ leads to an abrupt increase/decrease of beacon rate. This would cause transient channel congestion/accuracy reduction. In the simulator, through trial and error, we set this value at 0.45. After obtaining the new beacon rate value, we can determine the value of Beacon Interval Time

(BIT), enabling the next beacon to be scheduled in BIT seconds. Moreover, the value of required beacon rate depends upon the designed fuzzy inference system.

Algorithm 1 Beacon interval time adaptation

Initialize BR_c
if *Beacon message is received* **then**
  Find percentage of same directional neighbour vehicles
  Find its own emergency status
  Trigger Fuzzy Inference System
  get the value of BR_r
  $BR_n = BR_c + \gamma(BR_r - BR_c)$
  $BR_c = BR_n$
  $BIT = \frac{1}{BR_c}$
  Output the value of BIT
end if

V.1.3 Emergency Status of Vehicles

In a real heterogeneous vehicular environment, different kinds of vehicles, with different kinds of status, are communicating with one another. During unusual traffic conditions, some vehicles may travel on the road with emergency status (e.g. ambulance, fire truck, police car, or it can be any vehicle in an emergency situation such as failing brakes). These vehicles should diffuse their emergency status to their neighbours abruptly, and with a high degree of accuracy. Thus, increased beacon rate is very crucial for these types of vehicles, even under congested traffic conditions. These vehicles need to be able to inform neighbour vehicles to clear the road, with extra cooperative accuracy. On the other hand, normal vehicles follow their usual beaconing rate based on mobility characteristics.

V.1.4 Percentage of Directional Vehicles

We mentioned a vehicle parameter known as emergency status. Beaconing frequency control depends upon the vehicles current status and the traffic condition of neighbour vehicles. This section elaborates the latter (percentage of directional vehicles). Mobility characteristics like direction, velocity, and traffic density are very important parameters to consider when adapting beacon rate in VANET. The reasons of this are summarized as follows. First, vehicles on the road travel in constrained directions, thus vehicle beacon rate adaptation should take both directions into consideration. For instance, in a vehicular scenario with two way traffic, and vehicles moving in one direction have congested traffic conditions, they should reduce beacon rate, whereas vehicles moving in the other direction may vary their own beacon rate. Second, the velocity of vehicles and traffic density are implicitly interrelated to one another. This relationship is clearly known in traffic flow theory as in [260] Kerner states that the vehicles average velocity decreases as a result of increasing vehicular traffic density. Therefore, the percentage of vehicles traveling in the same direction is considered as an input as this parameter implicitly combines direction of vehicles, traffic density and velocity of vehicles.

In Figure A.17, vehicles 1, 2, 3, 4 and 5 are moving in the same direction, while 6, 7 and 8 are traveling in the opposite direction. If vehicle 1 wants to find the percentage of neighbor vehicles in the same direction, it can perform the following calculation:

$$PDN = \frac{NND}{TNN}$$

where PDN is the percentage of the same direction neighbor nodes, NND determines the number of the same direction neighbour nodes and TNN is the total number of neighbour nodes. Thus, the value of PDN for vehicle one is 0.5715, which means that this percentage of vehicles is moving in the same direction. In this way, this percentage implicitly considers combined direction, traffic density and velocity. Additionally, a vehicle can calculate its relative direction with other vehicles when its own and neighbours direction are known. For example: IF vehicle *a* is moving in *(dxa, dya)* direction and vehicle *b* is moving in *(dxb, dyb)* direction we can calculate the bearing angle *(σ)* between a vehicle and its neighbour as follows:

$$\cos\sigma = \frac{dx_a \cdot dx_b + dy_a \cdot dy_b}{\sqrt{dx_a^2 + dy_a^2} \cdot \sqrt{dx_b^2 + dy_b^2}}$$

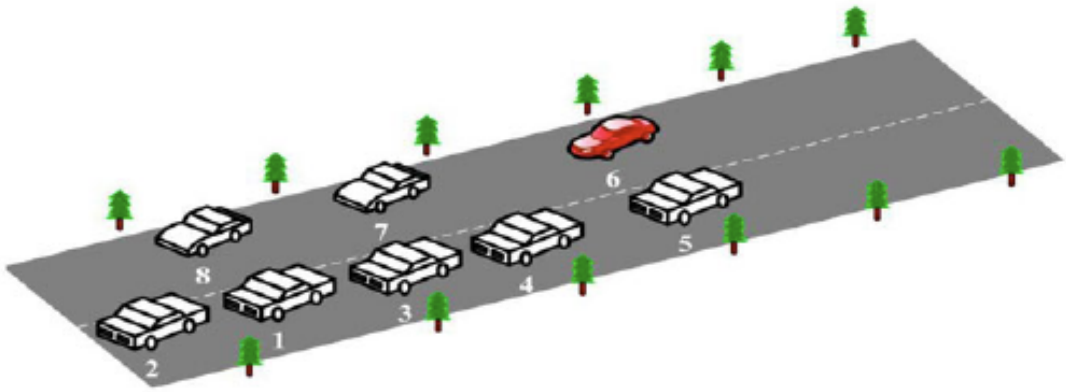


Figure A.17 Vehicular scenario

V.1.5 Fuzzification of inputs and outputs

The two input parameters to be fuzzified are the Percentage of Directional Neighbour Vehicles (PDN) and Vehicle Status (VS), as illustrated in Figure A.18. The membership functions named *Sparse*, *MDense* and *VDense* are used to represent the *PDN*. The selection of *PDN* membership functions can be derived based on experience as well as trial and error of the application requirement, thus the range begins at (0) and ends at (1). The reasoning behind this range is that a node might not have any same directional neighbour node (0) or all vehicles are moving in the same direction (1). When vehicles are in motion, the value of PDN may vary between its minimum and maximum value. Thus, the value of beacon rate is adapted in response to this percentage variation intelligently combined with the status of vehicles.

In addition, the VS fuzzy variable is represented as sharp/discrete values because status of vehicles is either emergency or non emergency. The discrete value representation of fuzzy variables is possible in fuzzy inference system. In our fuzzy inference system, we utilize the membership functions *Emerg* and *NEmerg* to represent the emergency/non emergency status of vehicles. As demonstrated in Fig. 3, there is no intersection between *Emerg* and *NEmerg* at the x-axes, thus it is a discrete representation of VS fuzzy variable. The output beacon rate is configured to a range between (1 to 10 beacon/second); the greater this value, the lower the duty cycle time for beacon generation. In addition, triangular functions are used as membership functions as they have been extensively used in real-time applications due to their simple formulas and computational efficiency. It is worth mentioning that the wise design of the membership function has a positive impact on the fuzzy decision making process performance.

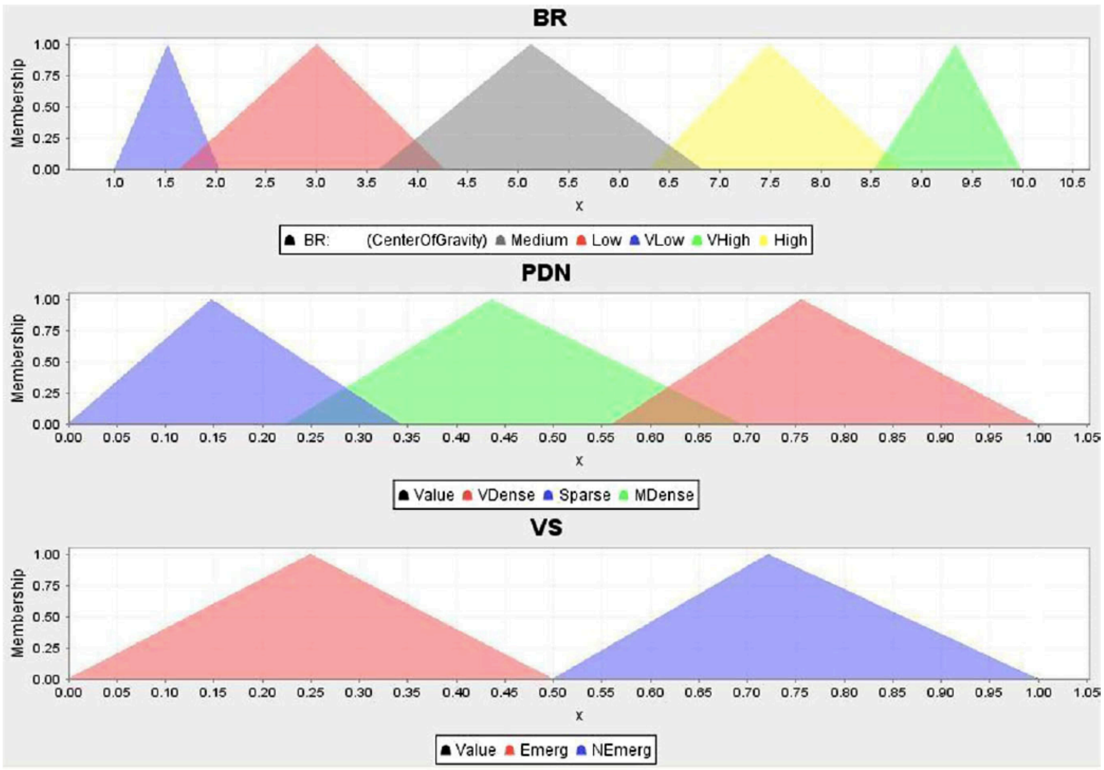


Figure A.18 Fuzzy membership functions for inputs (VS – Vehicle Status - and PDN – Percentage of Directional Neighbour vehicles -) and output (Beaconing Rate)

V.1.6 Fuzzy inference engine

The fuzzy inference engine is a group of rules developed using expert knowledge. We have designed the knowledge based rules that connect the inputs and the outputs based on a careful understanding of the philosophy behind vehicular network behaviour. The fuzzy inference system is designed based on 6 rules which are presented in the following table (knowledge structure).

Rule	IF		THEN
	Perce. of Direc.	Vehicle Status	BR_r
1	Sparse	Emerg.	VHigh
2	MDense	Emerg.	High
3	VDense	Emerg.	Medium
4	Sparse	NEmerg.	Medium
5	MDense	NEmerg.	Low
6	VDense	NEmerg.	VLow

In order to demonstrate the correct operation of our designed system one rule is used to show how the inference engine works and the outputs of each rule are combined for generating the fuzzy decision (Mamdani, E. H. (1977). Application of fuzzy logic to approximate reasoning using linguistic synthesis. *IEEE Transactions on Computers*). Consider a rule If (PDN is Sparse) and (VS is NEmerg) then (BR (beacon/second) is Medium) as an example of calculating output of the specified rule. In our fuzzy inference system, in the case where PDN is 0.206 and VS is 0.532, the beacon rate is 5.22 beacon/second.

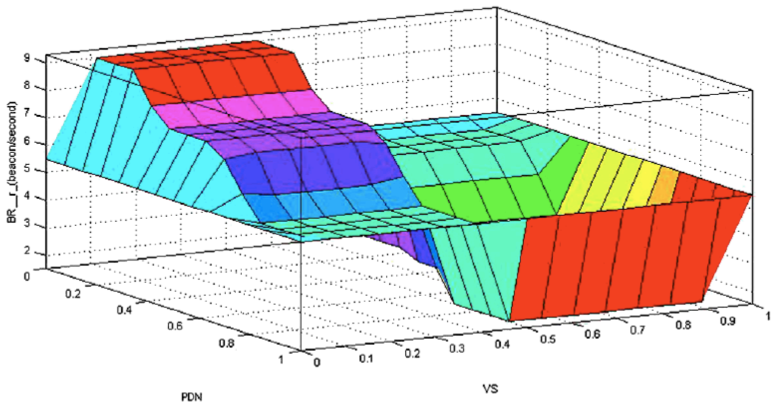


Figure A.19 Correlation between input and output

In order to calculate beaconing intervals based on Algorithm 1, let us assume that the value of *BRc* is 4.7 beacon/second and the output crisp value of fuzzy inference system for *BRr* is 5.22 beacon/second. The value of the new beacon rate (*BRn*) is equivalent to 4.934 beacon/second. After taking the reciprocal of *BRn*, the duty cycle of the new beacon interval becomes 0.2027 second. The vehicle has this beacon interval because of its non-emergency status and the sparse distribution of neighbour vehicles in the vicinity zone. It means our fuzzy inference system uses a tradeoff decision between parameters (VS and PDN) to adaptively tune the beacon rate. This output is obtained by using Mamdani's fuzzy inference method. Furthermore, Figure A.19 depicts the correlation behaviour between input and output variables. The trend shows that the value of output beacon rate increases when the value of PDN is between 0 to 0.2 as well as VS between 0 to 0.5. This is because of the emergency status of the vehicle and the lower percentage of directional neighbour vehicles (upper dark red part). Thus, our fuzzy inference system could increase beacon rate as traffic density decreases (velocity increases) or vice versa.

APPENDIX VI Acronyms

ACK	Acknowledgement
ACM	Association for Computing Machinery
AI	Artificial Intelligence
AM	Atomic Mechanism
AQoS	Adaptation Quality of Service
AS	Atomic Service
BB	Building Block
BER	Bit Error Rate
BGP	Border Gateway Protocol
BPEL	Business Process Execution Language
CBR	Condition Based Routing
CCN	Content-Centric Networking
CM	Content Managment
CPU	Central Processing Unit
CRC	Cyclic Redundancy Check
CREQ	Communication Request
CRESP	Communication Response
CRSV	Communication Reservation
CS	Composed Service
CSMA/CA	Carrier Sense Multiple Access with Collision Avoidance
DCF	Distributed Coordination Function
DHT	Distributed Hash Table
DNS	Domain Name Server
DONA	Data-Oriented Network Architecture
e2e	End-2-End
EDCA	Enhanched Distributed Channel Access
ESN	End Service Node
FEC	Forward Error Correction
FI	Future Internet
FIA	Future Internet Assembly
FIND	Future Internet Design
FN	Future Network
FW	Firewall
GENI	Global Environment for Network Innovations
GPS	Global Positioning System
HTTP	HyperText Transfer Protocol
ICN	Information-Centric Networking
ID	Identifier
IEEE	Institute of Electrical and Electronics Engineers
IMS	IP Multimedia Subsystem
IN	Intermediate Node
IP	Internet Protocol
IPTV	Television over IP
ISO	International Organization for Standardization

ITU	International Telecommunication Union
JPEG	Joint Photographic Experts Group
JXTA	Juxtapose
LAN	Local Area Network
LC	Layered Coding
LISP	Locator/ID Separator Protocol
MAC	Medium Access Control
MDC	Multiple Description Coding
MIT	Massachussets Institute of Technology
MPLS	Multiprotocol Label Switching
MVC	Multiview Video Coding
N3	Notation 3
NAT	Network Address Translator
NC	Network Coding
NE	Network Element
NG	Next Generation
NGN	Next Generation Network
Ninja SDS	Ninja Service Discovery Service
NQoS	Network Quality of Service
NS-2	Network Simulator 2
NSF	National Science Foundation
OASIS	Organization for the Advancement of Structured Information Standards
OMA	Open Mobile Alliance
OS	Operatying System
OTT	Over The Top
P2P	Peer-to-Peer
PC	Personal computer
PDA	Personal Digital Assistant
PEAQ	Perceptual Evaluation of Audio Quality
PEAVQ	Perceptual Evaluation of Video Quality
PN	Provider Node
PQoS	Perceived Quality of Service
PSIRP	Publish Subscribe Internet Routing
PSNR	Peak Signal to Noise Ratio
QoE	Quality of Experience
QoS	Quality of Service
RAM	Random Access Memory
RBA	Role-Based Architecture
RDV	Rendezvous
RGB	Red Green Blue
Rid	Resource Identifier
RINA	Recursive Inter-Network Architecture
RN	Requester Node
RNA	Recursive Network Architecture
RTCP	Real Time Control Protocol

RTP	Real-time Transport Protocol
SDP	Session Description Protocol
SDPng	Session Description Protocol Next Generation
Sid	Scope Identifier
SILO	Service Integration Control and Optimization
SIP	Session Initiation Protocol
SLA	Service Level Agreement
SNR	Signal to Noise Ration
SOA	Service-Oriented Architecture
SOAP	Simple Object Access Protocol
SOC	Service-Oriented Computing
SON	Service-Oriented Network
SONATE	Service-Oriented Network Architecture
SSIM	Structural Similarity
SVC	Scalable Video Coding
TARIFA	The Autonomic Redesign of the Internet Future Architecture
TCP	Transmission Control Protocol
TV	Television
UDDI	Universal Description Discovery Language
UDP	User Datagram Protocol
VANET	Vehicular Ad-hoc network
VoD	Video on Demand
WF	WorkFlow
Wifi	Wireless Fidelity
WSDL	Web Service Description Language
WWW	World Wide Web
XML	eXtensible Markup Language

APPENDIX VII Generated Publications

This appendix lists the publications and dissemination efforts associated with this PhD. Thesis. It includes international journal papers, conference papers (national and international), workshops, book chapters and international standards contributions.

- Book Chapters

[GP1] Book Publisher: River Publisher, Conference Name: Advances in Next Generation Services and Service Architectures (ANGSA), Volume I, Title: NGN-based subscriber's context processing telco service architecture: design and implementation, Authors: Alberto J. Gonzalez (UPC/i2CAT), Andre Rios (UPC), Jesus Alcober (UPC/i2CAT), Alejandro Cadenas (Telefonica R+D). ISBN: 978-87-92329-55-4.

[GP2] Book Publisher: River Publisher, Conference Name: Advances in Next Generation Services and Service Architectures (ANGSA), Volume II (Future Internet Services and Service Architectures), Title: Collaborative and Interactive Media Services in P2P Systems: Trends and Challenges, Authors: Andre Rios (UPC), Alberto J. Gonzalez (UPC/i2CAT), Jesus Alcober (UPC/i2CAT), Javier Ozon (UPC), Kayhan Z. Ghafoor (University of Malaysia). ISBN: 978-87-92329-59-2.

[GP3] A. Rios, A. J. González, and J. Alcober. "Peer-to-Peer Multimedia Conferencing System based on SIP Signaling". Traffic and Performance Engineering for Heterogeneous Networks, Three Research Volumes by River Publishers, February 2009. ISBN 978-87-92329-16-5.

- International Journals

[GP4] Kayhan Zrar Ghafoor, Kamalrulnizam Abu Bakar, Zaitul Zainuddin, C.-H. Ke and Alberto J. Gonzalez, "Reliable Video Geocasting over Vehicular Ad Hoc Networks", International Journal of Ad-Hoc & Sensor Wireless Networks, Vol. 15, Nr. 2-4, p201-221, 2012, http://oldcitypublishing.com/AHSWN/AHSWNcontents/AHSWNv15.2-4contents.html, ISSN: 1551-9899.

[GP5] Kayhan Zrar Ghafoor, Kamalrulnizam Abu Bakar, Martijn van Eenennaam, Rashid Hafeez Khokhar and Alberto J. Gonzalez, "A fuzzy logic approach to beaconing for vehicular ad hoc networks", International Journal of Telecommunication Systems, SpringerLink, 53 (4), 2013, http://dx.doi.org/10.1007/s11235-011-9466-8, 2011. ISSN: 1018-4864.

[GP6] Xavier Miguelez, Alberto J. Gonzalez, Francisco J. Iglesias, Jesus Alcober, "Flexible media transport framework based on service composition for Future Network", OSIA Journal Standard and Technology Review (http://www.osia.or.kr), March 2011, vol. 24, num. 1, p. 42-60, Korea, 2011.

- International Conferences

[GP7] Alberto J. Gonzalez, Ramon Martin de Pozuelo, Kayhan Zrar Ghafoor, Francesc Pinyol, Jesus Alcober, "Context-aware Multimedia Service Composition Using Quality Assessment", IEEE International Conference on Multimedia and Expo (ICME 2011), Barcelona, 11-15 July, 2011. ISBN 978-1-4244-4291-1.

[GP8] Alberto J. Gonzalez, Ramon Martin-de-Pozuelo, Alfredo Gutierrez, Jesus Alcober, Francesc Pinyol, Josep Maria Monguet, "Costing framework for service-oriented Future Internet architectures: empowering requester's choice", ACM 6th International Conference on Future Internet Technologies (CFI 2011), In cooperation with ACM SIGCOMM, Seoul, Korea, 2011. ISBN: 978-1-60558-686-1.

[GP9] Alberto J. Gonzalez, Kayhan Zrar Ghafoor, Ramon Piney, Andre Rios, Jesus Alcober, Kamalrulnizam Abu Bakar, "Fuzzy Redundancy Adaptation and Joint Source Network Coding for VANET Video Streaming", IFIP 9th International Conference on Wired/Wireless Internet Communications, WWIC 2011, Vilanova i la Geltrú, Spain, 2011. Lecture Notes in Computer Science, 2011, Volume 6649/2011, 458-469.

[GP10] Ali Safa Sadiq, Kamalrulnizam Abu Bakar, Kayhan Zrar Ghafoor and Alberto J. Gonzalez, "Mobility and Signal Strength- Aware Handover Decision in Mobile IP6 based Wireless LAN", IAENG International MultiConference of Engineers and Computer Scientists 2011 (IMECS 2011), Hong Kong, 16-18 March, 2011. (ISI/SCOPUS Cited Publication). [http://www.iaeng.org/publication/IMECS2011/IMECS2011_pp664-669.pdf]

[GP11] Kayhan Zrar Ghafoor, Kamalrulnizam Abu Bakar, Alberto J. Gonzalez, "Error Resilient Aware Video Streaming over Vehicular Ad Hoc Networks", In Proceedings of the International Conference on Postgraduate Education (ICPE-4), 2010, Kuala Lumpur, Malaysia.

[GP12] Alberto J. Gonzalez, Andre Rios, Guillermo Enero, Antoni Oller, Jesus Alcober, "Evaluating MDC with Incentives in P2PTV Systems", F.A. Aagesen and S.J. Knapskog (Eds.): EUNICE 2010, LNCS 6164, pp. 266--269. IFIP International Federation for Information Processing (2010). ISBN:3-642-13970-1 978-3-642-13970-3.

[GP13] Xavier Sanchez Loro, Alberto J. Gonzalez, Ramon Martin-de-Pozuelo, "A Semantic Context-Aware Network Architecture", Future Network & Mobile Summit 2010, Florence, Italy, June 2010. ISBN: 978-1-905824-16-8.

[GP14] Alberto Jose Gonzalez, Andre Rios and Jesus Alcober, "Robust and Scalable P2P Streaming for Future Media Internet", Future Internet Assembly. Stockholm, Kista Science City, Sweden, November 2009.

[GP15] A. Rios, F. Enrich, X. Milà, J. F. Crespo, A. González, and A. Vidal. "A Peer-to-Peer Live Streaming Platform for High-Quality Media Services". NEM Summit 2008 Towards Future Internet. Saint Malo, France. ISBN 978-3-00-025978-4, October 2008.

[GP16] A. Rios, A. J. González, A. Oller, J. López, and J. Alcober. "P2P Multipoint Conference System using SIP". HETNET 2008. Kalskrona, Sweden, February 2008.

- National Conferences

[GP17] Alberto José González, Francesc Rillo, Jesus Alcober, Daniel Rodríguez, Javier López. "Streaming P2P robusto en redes Ad-hoc utilizando información social", JITEL 2009, Cartagena, Spain, September 2009. ISBN: 978-84-96997-27-1.

[GP18] A. Ríos, A. J. González, and J. Alcober. "Prototype of P2P Multiconferencing System based on SIP". Telecom I+D 2008, Bilbao, Spain. ISBN 978-84-9860-135-0, October 2008.

- International Workshops

[GP19] Alberto J. Gonzalez, "Media Transport and Service composition in Future Network", Joint ITU-T SG 13 and ISO/IEC JTC 1/SC 6 workshop on "Future Networks standardization, Geneva, Switzerland, June 2012. (Invited)

[GP20] Alberto J. Gonzalez, "ISO/IEC JTC1 SC6 WG7 Future Network – Part 7: Service Composition", Future Internet Standards Workshop, Seoul, Korea, December 2011. (Invited)

[GP21] Alberto J. Gonzalez, Jesus Alcober, Ramon Martin-de-Pozuelo, Francesc Pinyol, "In-Network Service Selection and Composition Based on User and Network Context", Euro-NF International Workshop on Traffic and Congestion Control for the Future Internet, Volos, Greece, 2011.

[GP22] Alberto Jose Gonzalez, Ramon Martin de Pozuelo, Xavier Sanchez, "FCN Contributions to FIA Valencia 2010 Break Out Sessions", 1st Future Content Networks (FCN) Group Workshop, Brussels, Belgium, January 2010.

- International Standardization activities

[GP23] ISO/IEC JTC1/SC6 Working Group 7 "Future Network", part 7 "Service Composition", document 29181-7 Service Composition: Problem Statement and Requirements. Editors: Alberto J. Gonzalez (UPC, Spain), Ramon Martin de Pozuelo (URL, Spain), Yi Jong Hwa (ETRI, Korea). (Technical Report)

Bibliography

[1] Approaching the Zettabyte Era, by Cisco.
[2] Ken Chen, Bernhard Plattner, “Active Networks: Architecture and flexible applications”, Annals of Telecommunications, Springer Paris, vol. 59, no. 5, pp. 499-501, 2004.
[3] Anja Feldmann. 2007. Internet clean-slate design: what and why?. *SIGCOMM Comput. Commun. Rev.* 37, 3 (July 2007), 59-64.
[4] Entering the Zettabyte Era, by Cisco. http://www.cisco.com/en/US/solutions/collateral/ns341/ns525/ns537/ns705/ns827/VNI_Hyperconnectivity_WP.html
[5] “Why do we need a Content Centric Future Internet? Proposal towards Content-Centric Internet architectures”, FIA-Future Content Networks Group, Prague, May, 2009
[6] V. Jacobson, D. K. Smetters, J. D. Thornton, M. F. Plass, N. H. Briggs, R. L. Braynard (PARC), “Networking Named Content”, CoNEXT 2009, Rome, December, 2009.
[7] D. Perino, M. Varvello, “A reality check for content centric networking”, ACM Sigcomm Workshop on Information-Centric Networking (ICN) 2011.
[8] TARIFA project website, www.i2cat.net/es/projecte/tarifa
[9] Next Media, Future Media Internet Coordination Action, Deliverable D.2.1 “Report on current research and business targets”, http://www.gatv.ssr.upm.es/nextmedia/images/deliverables/nextmedia_d2.1_syn_v1.0_20100311.pdf
[10] “Research on Future Media Internet”, Future Media Internet Task Force, ftp://ftp.cordis.europa.eu/pub/fp7/ict/docs/netmedia/research-on-future-media-internet-2009-4072_en.pdf
[11] PPLive official site, www.pplive.com
[12] PPStream official site, www.ppstream.com
[13] UUSee official site, www.uusee.com
[14] Pando official site, www.pando.com
[15] Zattoo official site, http://zattoo.com/
[16] Youtube official site, www.youtube.com
[17] Cachelogic official site, www.cachelogic.com/
[18] Bitttorrent official site, www.bittorrent.com/
[19] Internet Study 2008-2009 by Ipoque, http://www.ipoque.com/resources/internet-studies
[20] CISCO IP traffic forecast, http://newsroom.cisco.com/dlls/2009/prod_060909.html
[21] M. van der Schaar and P. Chou, “Multimedia over IP and Wireless Networks: Compression, Networking, and Systems” 2007.
[22] A. Ríos, A. J. González, and J. Alcober. “Prototype of P2P Multiconferencing System based on SIP”. Telecom I+D 2008, Bilbao, Spain (Local Conference in Spanish). ISBN 978-84-9860-135-0, October 2008.
[23] Alberto J. Gonzalez, “Robust P2P Live Streaming”, Master Thesis (2009).
[24] T. Kathiravelu and A. Pears. “Approaches to P2P Internet Application Development”. ICAS/ICNS 2005, October 2005, Papeete, Tahiti.
[25] Y. Chu, S. G. Rao, and H. Zhang, “A case for end system multicast,” ACM SIGMETRICS’00, June 2000.
[26] S. Banerjee, B. Bhattacharjee, and C. Kommareddy. “Scalable application layer multicast”. ACM SIGCOMM’02, Pittsburgh, PA, Aug. 2002.
[27] S. Ratnasamy, P. Francis, M. Handley, R. Karp, and P. Shenker. “A scalable content-ddressable network”. ACM SIGCOMM 2001, August 2001.
[28] I. Stoica, R. Morris, D. Karger, and H. Balakrishnan. “Chord : A scalable peer-to-peer lookup service for internet applications”. ACM SIGCOMM 2001 Technical Conference, August 2001.
[29]] A. Rowston and P. Druschel. “Pastry : Scalable decentralized object location and routing and for large scale peer-to-peer systems”. FIP/ACM Middleware 2001, November 2001.
[30] B. Y. Zhao, J. D. Kubiatowicz, and A. D. Joseph. “Tapestry: An infrastructure for fault-tolerant wide-area location and routing”. Technical Report UCB-CSD-01-1141, UC Berkeley, April 2001.

[31] V. Pai, K. Kumar, K. Tamilmani, V. Sambamurthy, and A. Mohr, Chainsaw: “Eliminating Trees from Overlay Multicast” *Lecture Notes in Computer Science*, vol. 3640, p. 127, 2005.

[32] X. Zhang, J. Liu, B. Li, and Y.-S. Yum, “CoolStreaming/DONet: adata-driven overlay network for peer-to-peer live media streaming” *INFOCOM 2005. 24th Annual Joint Conference of the IEEE Computer and Communications Societies. Proceedings IEEE*, vol. 3, pp. 2102-2111, vol. 3, March 2005.

[33] L. Zhao, J.-G. Luo, M. Zhang, W.-J. Fu, J. Luo, Y.-F. Zhang, and S.-Q.Yang, “Gridmedia: A Practical Peer-to-Peer Based Live Video Streaming System” *Multimedia Signal Processing*, 2005 IEEE 7th Workshopon, pp. 1-4, Oct. 2005.

[34] F. Picconi and L. Massoulié, “Is There a Future for Mesh-Based liveVideo Streaming?” *in Proceedings of the 2008 Eighth International Conference on Peer-to-Peer Computing-Volume 00*, pp. 289-298, IEEE Computer Society Washington, DC, USA, 2008.

[35] Y. Guo, C. Liang, and Y. Liu, “AQCS: Adaptive Queue-Based Chunk Scheduling for P2P Live Streaming” *Lecture Notes in Computer Science*, vol. 4982, p. 433, 2008.

[36] Jose Francisco Crespo, “CoolRuc 2.0: a P2P Live Streaming using Quality-Adaptable Video Codings”, Master Thesis (2009).

[37] T. Koponen,M. Chawla, B.-G. Chun, A. Ermolinskiy, K. H. Kim, S. Shenker, I. Stoica, “A Data-Oriented (and Beyond) Network Architecture”, in Proceedings of SIGCOMM, 2007.

[38] SIENA Abstract, http://www.isr.uci.edu/events/twist/wisen98/abstracts/abs_carzaniga.html

[39] Palo Alto Research Center content-centric network, http://www.parc.com/ research/ projects/networking/contentcentric/

[40] Open CDN project http://labtel.ing.uniroma1.it/opencdn/

[41] Oscar-Data-oriented Structured Overlay http://www.ntu.edu.sg/home/ anwitaman/ ResearchOscar.html

[42] Sethuram Kodeswaran et al., "Content and Context Aware Networking (COCONET) Using Semantic Tagging", International Workshop on Semantics enabled Networks and Services, April 2006.

[43] AKAMAI, http://www.akamai.com/html/technology/edgeplatform.html

[44] V. K. Goyal. ”Multiple Description Coding: Compression Meets the Network”. IEEE Signal Processing Magazine. September 2001.

[45] H. Schwarz, D. Marpe and T. Wiegand, ―Overview of the scalable video coding extension of the H.264 / AVC standard,‖ IEEE Transactions on Circuits and Systems for Video Technology, Vol. 17, Iss. 9, pp. 1103 - 1120, September (2007).

[46] M. Mrak, N. Sprljan, T. Zgaljic, N. Ramzan, S. Wan and E. Izquierdo, Performance evidence of software proposal for Wavelet Video Coding Exploration group, ISO/IEC JTC1/SC29/WG11/ MPEG2006/M13146, 76th MPEG Meeting, Montreux, Switzerland, April (2006).

[47] M. Wien, H. Schwarz and T. Oelbaum, ―Performance analysis of SVC, IEEE Transactions on Circuits and Systems for Video Technology, Vol. 17, Iss. 9, pp. 1194 – 1203, September (2007).

[48] A. Vetro, P. Pandit, Kimata, H., Smolic, A.: Joint draft 8.0 on multiview video coding. Joint Video Team (JVT) of ISO/IEC MPEG ITU-T VCEG ISO/IEC JTC1/SC29/WG11 and ITU-T SG16 Q.6 (2007).

[49] Survey of algorithms used for multi-view video coding (mvc). ISO/IECJTC1/SC29/WG11 Doc N6909 (2005).

[50] Smolic, A., Kimata, H.: Report on 3dav exploration. ISO/IEC JTC1/SC29/WG11 Doc N5878 (2003).

[51] C. Bilen, A. Aksay, Bozdagi Akar, G.: A multi-view video codec based on H.264. In: Proc. IEEE Conf. Image Proc. (ICIP), Oct. 8-11, Atlanta, USA (2006).

[52] C. Fragouli and E. Soljanin. ”Network Coding Fundamentals (Foundations and Trends in Networking)”. Now Publishers Inc. June 10, 2007.

[53] C. Fragouli and E. Soljanin.”Network Coding Applications”. Now Publishers Inc. January 8, 2008.

[54] F. Pereira, J.R. Smith, A. Vetro, (eds.), "Special Section on MPEG-21", IEEE Transactions on Multimedia, vol. 7, no. 3, Jun. 2005.

[55] Th. Zahariadis, C. Lamy-Bergot, Th. Schierl, K. Grüneberg, L. Celetto, C. Timmerer, “Content Adaptation Issues in the Future Internet,” IOS press ―Towards the Future Internet - A European Research Perspective,” 2009, ISBN 978-1-60750-007-0 pp. 283-292.

[56] Th. Zahariadis, O. Negru, F. Rovati, F. Álvarez, "Seamless Content Delivery over the Future Internet" TEMU 2008, 16-18 July 2008.
[57] M. N. Huhns, M. P. Singh, Service-oriented computing: Key concepts and principles, IEEE Internet Computing 9 (1) (2005) 7581.
[58] M. P. Papazoglou, P. Traverso, S. Dustdar, F. Leymann, Service- oriented computing: State of the art and research challenges, Computer-IEEE Computer Society. 40 (11) (2007) 38-45.
[59] K. M. Gnschka, L. Froihofer, S. Dustdar, What SOA can do for software dependability, Supplementary Volume of DSN 8.
[60] D. Mennie, B. Pagurek, An architecture to support dynamic composition of service components, Systems and Computer Engineering. Carleton University, Canada.
[61] R. Kumar, Towards a relationship oriented service architecture.
[62] P. Lalanda, C. Marin, A domain-con_gurable development environment for service-oriented applications, IEEE SOFTWARE (2007) 3138.
[63] D. S. Coalition, A. Ankolekar, et al., DAML-S: web service description for the semantic web, in: First International Semantic Web Conference, Sardinia, Italy, 2002.
[64] D. Kuropka, H. Meyer, Survey on Service Composition, Univ.-Verl., 2005.
[65] G. Yang, M. Kifer, H. Wan, C. Zhao, FLORA-2 manual, http://fora.sourceforge.net/ (2008).URL http://flora.sourceforge.net/
[66] D. Gelernter and N. Carriero, "Coordination languages and their significance," *Communications of the ACM*, vol. 35, 1992, pp. 97–107.
[67] S.R. Ponnekanti and A. Fox, "Sword: A developer toolkit for web service composition," *Proc. of the Eleventh International World Wide Web Conference, Honolulu, HI*, 2002.
[68] D.W. Evren, D. Wu, E. Sirin, J. Hendler, D. Nau, and B. Parsia, "Automatic Web Services Composition Using SHOP2," *In Workshop on Planning for Web Services*, 2003.
[69] D. Berardi, D. Calvanese, G. De Giacomo, and M. Mecella, "Reasoning about actions for e-service composition," *Proceedings of ICAPS*.
[70] S. Thakkar, C.A. Knoblock, J.L. Ambite, and others, "A view integration approach to dynamic composition of web services," *Proceedings of 2003 ICAPS Workshop on Planning for Web Services*, 2003.
[71] S.K. Garg and R.B. Mishra, "TRS: system for recommending semantic web service composition approaches," *International Symposium on Information Technology, 2008. ITSim 2008*, 2008, pp. 1–5.
[72] W3C, Web services architecture, http://www.w3.org/TR/ws-arch/ (2004).
[73] F. Curbera, M. Duftler, R. Khalaf, W. Nagy, N. Mukhi, and S. Weerawarana, "Unraveling the Web services web: an introduction to SOAP, WSDL, andUDDI," *IEEE Internet computing*, vol. 6, 2002, pp. 86–93.
[74] S. Thakkar, C.A. Knoblock, J.L. Ambite, and others, "A view integration approach to dynamic composition of web services," *Proceedings of 2003 ICAPS Workshop on Planning for Web Services*, 2003.
[75] M. Vukovic and P. Robinson, "Context aware service composition," Proceedings of the Third UK UbiNet Workshop, Bath, UK, 2005.
[76] A. Alamri, M. Eid, and A. El Saddik, "Classification of the state-of-the-art dynamic web services composition techniques," International Journal of Web and Grid Services, vol. 2, 2006, pp. 148–166.
[77] V. Agarwal, G. Chafle, S. Mittal, and B. Srivastava, "Understanding approaches for web service composition and execution," Proceedings of the 1st Bangalore annual Compute conference, 2008, p. 1.
[78] K. Sivashanmugam, J.A. Miller, A.P. Sheth, and K. Verma, "Framework for semantic web process composition," International Journal of Electronic Commerce, vol. 9, 2005, pp. 71–106.
[79] S. McIlraith and T.C. Son, "Adapting golog for composition of semantic web services," PRINCIPLES OF KNOWLEDGE REPRESENTATION AND REASONING-INTERNATIONAL CONFERENCE-, 2002, pp. 482–496.
[80] H.J. Levesque, R. Reiter, Y. Lesperance, F. Lin, and R.B. Scherl, "GOLOG: A logic programming language for dynamic domains,"The Journal of Logic Programming, vol. 31,1997, pp. 59–83.
[81] S.Y. Su, J. Meng, R. Krithivasan, S. Degwekar, and S. Helal, "Dynamic inter-enterprise workflow management in a constraint-based e-service infrastructure," Electronic commerce research, vol. 3, 2003, pp. 9–24.
[82] S. Dustdar and W. Schreiner, "A survey on web services composition," International Journal

of Web and Grid Services, vol. 1, 2005, pp. 1–30.
[83] N. Channa, S. Li, A.W. Shaikh, and X. Fu, "Constraint satisfaction in dynamic web service composition," Sixteenth International Workshop on Database and Expert Systems Applications, 2005. Proceedings, 2005, pp. 658–664.
[84] D. VanderMeer, A. Datta, C. Technologyies, K. Dutta, H. Thomas, K. Ramamritham, and S.B. Navathe, "FUSION: A system allowing dynamic web service composition and automatic execution."
[85] B. Benatallah, Q.Z. Sheng, A.H. Ngu, and M. Dumas, "Declarative composition and peer-to-peer provisioning of dynamic web services," icde, 2002, p. 0297.
[86] S. Ambroszkiewicz, "enTish: An Approach to Service Description and Composition," In Proceedings of the First European Workshop on Object Orientation and Web Services, S, pp. 49–53.
[87] S.J. Russell and P. Norvig, Artificial intelligence: a modern approach, Prentice hall, 2009.
[88] S. McIlraith and R. Fadel, "Planning with complex actions," Workshop Program, p. 13.
[89] M. Ghallab, D.S. Nau, and P. Traverso, Automated Planning: theory and practice, Morgan Kaufmann Publishers, 2004.
[90] U. Küster, M. Stern, and B. König-Ries, "A classification of issues and approaches in automatic service composition," Intl. Workshop WESC, 2005.
[91] M. Bhatt, A. Flahive, C. Wouters, W. Rahayu, D. Taniar, and T. Dillon, "A distributed approach to sub-ontology extraction."
[92] M. Bhatt, C. Wouters, A. Flahive, W. Rahayu, and D. Taniar, "Semantic completeness in sub-ontology extraction using distributed methods," Computational Science and Its Applications–ICCSA 2004, 2004, pp. 508–517.
[93] A. Flahive, W. Rahayu, D. Taniar, and B. Apduhan, "A distributed ontology framework for the grid," Parallel and Distributed Computing: Applications and Technologies, 2005, pp. 68–71.
[94] M. Mrissa, D. Benslimane, Z. Maamar, and C. Ghedira, "Towards a semantic-and context-based approach for composing web services," International Journal of Web and Grid Services, vol. 1, 2005, pp. 268–286.
[95] C. Pahl, "A conceptual architecture for semantic web services development and deployment," International Journal of Web and Grid Services, vol. 1, 2005, pp. 287–304.
[96] K. Fujii and T. Suda, "Dynamic service composition using semantic information," Proceedings of the 2nd international conference on Service oriented computing, 2004, pp. 39–48.
[97] E. Sirin, J. Hendler, and B. Parsia, "Semi-automatic composition of web services using semantic descriptions," Web Services: Modeling, Architecture and Infrastructure workshop in conjunction with ICEIS2003, 2003.
[98] J. Vogel, L. Xu, F. Schnabel, R. Krummenacher, S. Abels, R. Gonzalez, A. Mos, N. Mehandjiev, L. Cekov, G. Ripa, and others, "State of the Art Report and Requirements for Service Construction. D6. 1.1, SOA4All EU project FP7-ICT-2008-215219," 2009.
[99] J.E. Cook and A.L. Wolf, "Software process validation: quantitatively measuring the correspondence of a process to a model," ACM Transactions on Software Engineering and Methodology (TOSEM), vol. 8, 1999, p. 176.
[100] R. Agrawal, D. Gunopulos, and F. Leymann, "Mining process models from workflow logs," Advances in Database Technology—EDBT'98, 1998, pp. 467–483.
[101] E.M. Gold, "Complexity of automaton identification from given data*," Information and control, vol. 37, 1978, pp. 302–320.
[102] R. Silva, J. Zhang, and J.G. Shanahan, "Probabilistic workflow mining," Proceedings of the eleventh ACM SIGKDD international conference on Knowledge discovery in data mining, 2005, p. 284.
[103] G. Greco, A. Guzzo, L. Pontieri, and D. Sacca, "Mining expressive process models by clustering workflow traces," Advances in Knowledge Discovery and Data Mining, 2004, pp. 52–62.
[104] A. K. Dey, Understanding and using context, Personal and ubiquitous computing 5 (1) (2001) 47.
[105] R. Ocampo, L. Cheng, Z. Lai, A. Galis, ContextWare support for network and service composition and self-adaptation, Mobility Aware Technologies and Applications (2005) 8495.
[106] M. Autili, V. Cortellessa, A. D. Marco, P. Inverardi, A conceptual model for adaptable context-

aware services, 2006, p. 1533.
[107] N. Baker, M. Zafar, B. Moltchanov, M. Knappmeyer, Context-Aware systems and implications for future internet, Towards the Future Internet (2009) 335.
[108] J. Domingue, D. Fensel, and R. González-Cabero, "SOA4All, enabling the SOA revolution on a world wide scale," *2008 IEEE International Conference on Semantic Computing*, 2008, pp. 530–537.
[109] L. Völker, D. Martin, I. El Khayat, C. Werle, and M. Zitterbart, "An Architecture for Concurrent Future Networks," *2nd GI/ITG KuVS Workshop on The Future Internet. Karlsruhe, Germany: GI/ITG Kommunikation und Verteilte Systeme*, 2008.
[110] B. Reuther, P. Müller: Future Internet Architecture - A Service Oriented Approach. In it - Information Technology Jahrgang 50 (2008) Heft 6, S. 383-389, Oldenbourg Verlag, Muenchen, 2008.
[111] D. Schwerdel, Z. Dimitrova, A. Siddiqui, B. Reuther, and P. Müller, "Composition of Self Descriptive Protocols for Future Network Architectures," 2009.
[112] L. Volker, C. Werle, and M. Zitterbart, "Decision process for automated selection of security protocols," *33rd IEEE Conference on Local Computer Networks, 2008. LCN 2008*, 2008, pp. 223–229.
[113] R.L. Keeney and H. Raiffa, "*Decisions with multiple objectives: Preferences and value tradeoffs"*, Cambridge University Press, 1993.
[114] R. Braden, T. Faber, and M. Handley, "From protocol stack to protocol heap: role-based architecture," ACM SIGCOMM Computer Communication Review, vol. 33, no. 1, pp. 1722, 2003.
[115] X. Sanchez-Loro, J. Casademont, J. Paradells, J.L. Ferrer, and A. Vidal, "Proposal of a clean slate network architecture for ubiquitous services provisioning," in Future Information Networks, 2009. ICFIN 2009. First International Conference on, 2009, pp. 54–60.
[116] John Day, Patterns in Network Architecture: A Return to Fundamentals, 2008.
[117] O. Sole Molina, "Multimedia quality assessment," M.S. thesis, Universitat Politecnica de Catalunya, 2009, directed by Alberto J. Gonzalez, http://hdl.handle.net/2099.1/7754
[118] "Future Media Internet Research Challenges and the Road Ahead", Future Media Internet Task Force, April 2010.
[119] IETF draft Session Description and Capability Negotiation draft-ietf-mmusic-sdpng-08, https://datatracker.ietf.org/doc/draft-ietf-mmusic-sdpng/
[120] ISO MPEG-21 overview, http://mpeg.chiariglione.org/standards/mpeg-21/mpeg-21.htm
[121] Gonzalo Camarillo, Miguel-Angel García-Martín, "The 3G IP Multimedia Subsystem (IMS): Merging the Internet and the Cellular Worlds", John Wiley & Sons, 2006, ISBN 0-470-01818-6.
[122] Mohammad Ilyas (Editor), Syed A. Ahson (Editor), "IP Multimedia Subsystem (IMS) Handbook", CRC Press, 2008, ISBN 1-4200-6459-2.
[123] ITU-T Study group 13, NGN Working group, http://www.itu.int/ITU-T/studygroups/com13/ngn2004/working_definition.html
[124] ETSI TISPAN NGN definition, http://www.etsi.org/tispan/
[125] Network Coding Theory (a tutorial), http://iest2.ie.cuhk.edu.hk/~whyeung/publications/tutorial.pdf
[126] Simon Dobson, Spyros Denazis, Antonio Fernández, Dominique Gaïti, Erol Gelenbe, Fabio Massacci, Paddy Nixon, Fabrice Saffre, Nikita Schmidt and Franco Zambonelli, "A Survey of Autonomic Communications. ACM Transactions on Autonomous and Adaptive Systems", 1(2):223-259, December 2006.
[127] Christos Gkantsidis and Pablo Rodriguez, "Network Coding for Large Scale Content Distribution", in *IEEE INFOCOM*, March 2005.
[128] Christos Gkantsidis, John Miller, and Pablo Rodriguez, "Comprehensive view of a live network coding P2P system", in *IMC*, Association for Computing Machinery, Inc., October 2006.
[129] Xavier Sanchez-Loro, Josep Paradells, Jordi Casademont, and José Luís Ferrer, "Optimizing the delivery chain in heterogenous networks", In *Proceedings of the 9th IFIP TC 6 international conference on Wired/wireless internet communications* (WWIC'11), Spain, 2011.
[130] C. Henke, A. Siddiqui, and R. Khondoker, "Network functional composition: State of the art," in ATNAC'10: Proceedings of Australasian Telecommunication Networks and Application Conference. IEEE, 2010.

[131] Xavier Sanchez Loro, Alberto J. Gonzalez, Ramon Martin-de-Pozuelo, “A Semantic Context-Aware Network Architecture”, Future Network & Mobile Summit 2010, Florence, Italy, June 2010. ISBN: 978-1-905824-16-8.
[132] Mea Wang. 2008. *A Quest for High-Performance Peer-To-Peer Live Multimedia Streaming*. Ph.D. Dissertation. University of Toronto, Toronto, Ont., Canada, Canada.
[133] V. Srivastava, M. Motani, Cross-layer design: a survey and the road ahead, Communications Magazine, IEEE. 43 (2005) 112–119.
[134] A.J. Gonzalez, R. Martin de Pozuelo, J. Alcober, F. Pinyol, A. Gutierrez, J.M. Monguet, Costing framework for service-oriented Future Internet architectures: empowering requester’s choice, in: Proceedings of the 6th International Conference on Future Internet Technologies, 2011: pp. 84–90.
[135] T. Anderson et al. “Overcoming Barriers to Disruptive Innovation in Networking”. GENI NSF Workshop Report, 2005.
[136] D. Raychaudhuri et al. “New Architectures and Disruptive Technologies for the Future Internet: The Wireless, Mobile and Sensor Network Perspective”. GENI NSF WMPG Report, August 2005.
[137] NewArch Project: Future-Generation Internet Architecture. http://www.isi.edu/newarch/
[138] D. Stevenson, R. Dutta, G. Rouskas, D. Reeves and I. Baldine. “On the Suitability of Composable Services for the Assurable Future” Milcom '07 October 2007, Orland FL.
[139] The FP7 4WARD Project. http://www.4ward-project.eu/
[140] Touch, J., Wang. Y., and Pingali, V., "A Recursive Network Architecture," ISI Technical Report ISI-TR-2006-626, October 2006.
[141] GENI.net Global Environment for Network Innovations. http://www.geni.net/
[142] NSF NeTS FIND Initiative. http://www.nets-find.net/index.php
[143] AKARI Project. http://akari-project.nict.go.jp/
[144] R. Dutta, G.N. Rouskas, I. Baldine, A. Bragg, D. Stevenson, The SILO architecture for services integration, control, and optimization for the future internet, in: Communications, 2007. ICC’07. IEEE International Conference on, nd: pp. 1899–1904.
[145] Fred Goldstein and John Day, “Moving beyond TCP/IP”, http://pouzin.pnanetworks.com/images/PSOCMovingBeyondTCP.pdf
[146] J.W. Kim, S.W. Han, D.H. Yi, N. Kim, C.C.J. Kuo, Media-oriented service composition with service overlay networks: Challenges, approaches and future trends, Journal of Communications. 5 (2010) 374–389.
[147] J.-W. Park, D. Yi, J. Kim, A futuristic service framework for flexible media-oriented service composition in future internet, in: ACM Press, 2010: p. 68.
[148] International Organization for Standardization (ISO), Standards development - Technical committees, http://www.iso.org/iso/standards_development/technical_committees.htm, (2011).
[149] ITU Telecommuncation Standardization Sector (ITU-T) Study Group 13, SIDE NGN - Draft Recommendation ITU-T Y.2240 (Y.NGN-SIDE-Req), Requirements and capabilities for NGN service integration and delivery environment, http://www.itu.int/md/meetingdoc.asp?lang=en&parent=T09-SG13-110117-TD&question=Q3/13, (2011).
[150] TeleManagement Forum (TM Forum), http://www.tmforum.org/browse.aspx, (2011).
[151] IEEE Standards Association, Next Generation Service Overlay Network (NGSON) Working Group, http://grouper.ieee.org/groups/ngson/, (2011).
[152] Future Internet Assembly (FIA), Future Internet, http://www.future-internet.eu/home/future-internet-assembly.html, (2011).
[153] Internet Research Task Force (IRTF) - Routing Research Group (RRG), http://trac.tools.ietf.org/group/irtf/trac/wiki/RoutingResearchGroup, (2011).
[154] D. Meyers, Report from the IAB Workshop on Routing and Addressing, Internet Draft, draft-iab-raws-report-01. (2007).
[155] D. Farinacci, Locator/ID separation protocol (LISP), Internet-draft, draft-farinacci-lisp-00. (2007).
[156] B. Quoitin, L. Iannone, C. de Launois, O. Bonaventure, Evaluating the benefits of the locator/identifier separation, in: Proceedings of 2nd ACM/IEEE international workshop on Mobility in the evolving internet architecture, 2007: p. 5.
[157] T. Koponen, M. Chawla, B.G. Chun, A. Ermolinskiy, K.H. Kim, S. Shenker, et al., A data-oriented (and beyond) network architecture, ACM SIGCOMM Computer Communication

Review. 37 (2007) 181–192.
[158] S. Tarkoma, M. Ain, K. Visala, The Publish/Subscribe Internet Routing Paradigm (PSIRP): Designing the Future Internet Architecture, Towards the Future Internet. (2009) 102.
[159] F. Zhu, M.W. Mutka, L.M. Ni, Service discovery in pervasive computing environments, Pervasive Computing, IEEE. 4 (2005) 81–90.
[160] F. Curbera, M. Duftler, R. Khalaf, W. Nagy, N. Mukhi, S. Weerawarana, Unraveling the Web services web: an introduction to SOAP, WSDL, and UDDI, IEEE Internet computing. 6 (2002) 86–93.
[161] C. Lee, S. Helal, Protocols for service discovery in dynamic and mobile networks, International Journal of Computer Research. 11 (2002) 1–12.
[162] G.G. Richard III, Service advertisement and discovery: enabling universal device cooperation, Internet Computing, IEEE. 4 (2000) 18–26.
[163] S. Helal, Standards for service discovery and delivery, Pervasive Computing, IEEE. 1 (2002) 95–100.
[164] R. Ahmed, N. Limam, J. Xiao, Y. Iraqi, R. Boutaba, Resource and service discovery in large-scale multi-domain networks, IEEE Communications Surveys & Tutorials. 9 (2007) 2–30.
[165] C. Lee, S. Helal, Context attributes: an approach to enable context-awareness for service discovery, (2003).
[166] G. Chen, D. Kotz, D.C.H.N.D.O.C. SCIENCE, Context-sensitive resource discovery, Citeseer, 2003.
[167] S. Bluetooth, Specification of the Bluetooth system, Core, version. 1 (2005) 2005–10.
[168] M.K. Bergman, The deep web: Surfacing hidden value, Journal of Electronic Publishing. 7 (2001) 07–01.
[169] S. Helal, N. Desai, V. Verma, C. Lee, Konark-a service discovery and delivery protocol for ad-hoc networks, in: Wireless Communications and Networking, 2003. WCNC 2003. 2003 IEEE, 2003: pp. 2107–2113.
[170] A.N. Mian, R. Baldoni, R. Beraldi, A survey of service discovery protocols in multihop mobile ad hoc networks, Pervasive Computing, IEEE. 8 (2009) 66–74.
[171] W. Bethea, R. Cole, P. Harshavardhana, Automated discovery of information services in heterogeneous distributed networks, in: Military Communications Conference, 2008. MILCOM 2008. IEEE, 2008: pp. 1–6.
[172] T. Gagnes, T. Plagemann, E. Munthe-Kaas, A conceptual service discovery architecture for semantic web services in dynamic environments, (2006).
[173] T. Gagnes, Assessing dynamic service discovery in the network centric battlefield, in: Military Communications Conference, 2007. MILCOM 2007. IEEE, nd: pp. 1–7.
[174] Y.D. Bromberg, V. Issarny, P.G. Raverdy, Interoperability of service discovery protocols: Transparent versus explicit approaches, Proceedings of the 15th IST Mobile & Wireless Communications Summit, Mykonos, June. (2006).
[175] J. Allard, V. Chinta, S. Gundala, G. Richard III, Jini meets UPnP: an architecture for Jini/UPnP interoperability, (2003).
[176] S. Kang, S. Ryu, N. Kim, Y. Lee, D. Lee, K.D. Moon, An architecture for interoperability of service discovery protocols using dynamic service proxies, Information Networking. Convergence in Broadband and Mobile Networking. (2005) 786–795.
[177] Xiaojun, H., Chao, L., Jian, L., Yong, L., Ross, K.: A Measurement Study of a Large-Scale P2P IPTV System. Multimedia, IEEE Transactions on, vol.9, no.8, pp.1672-1687 (2007)
[178] Chakareski, J., Han, S., Girod, B.: Layered coding vs. multiple descriptions for video streaming over multiple paths. In Proc. of the Eleventh ACM international Conference on Multimedia. Berkeley, CA, USA (2003).
[179] Soldani, C., Leduc, G., Verdicchio, F., Munteanu, A.: Multiple Description Coding versus Transport Layer FEC for Resilient Video Transmission. Digital Telecommunication, 2006. ICDT '06. International Conference on vol., no., pp.20-20, 29-31 (2006).
[180] P2PTVSIM, http://www.napa-wine.eu/cgi-bin/twiki/view/Public/P2PTVSim
[181] Lobb, R., Couto da Silva, A., Leonardi, E., Mellia, M., Meo, M.: Adaptive Overlay Topology for Mesh-Based P2P-TV Systems. In Proc. of ACM NOSSDAV 2009 (2009).
[182] Naicken, S., Basu, A., Livingston, B., Rodhetbhai, S.: A survey of peer-to-peer network simulators. In Proc. of the 7th Annual Postgraduate Symposium (PGNet '06) (2006).
[183] Seibert, J., Zage, D., Fahmy, S., Nita-Rotaru, C.: Experimental comparison of peer-to-peer streaming overlays: An application perspective. Local Computer Networks, LCN 2008. 33rd IEEE Conference on , vol., no., pp.20-27, 14-17 (2008).

[184] Habib, A., Chuang, J.: Incentive mechanism for peer-to-peer media streaming. Quality of Service, IWQOS 2004. Twelfth IEEE International Workshop on , vol., no., pp. 171-180, 7-9 (2004).
[185] Liu, Z., Shen, Y., Panwar, S.S., Ross, K., Wang, Y.: Using layered video to provide incentives in P2P live streaming. In Proc. of the 2007 Workshop on Peer-To-Peer Streaming and IP-TV (Kyoto, Japan, August 27 - 31, 2007). P2P-TV '07. ACM, New York, NY, 311-316 (2007).
[186] Zhengye, L., Yanming, S., Panwar, S.S., Ross, K.W., Wang, Y.: P2P Video Live Streaming with MDC: Providing Incentives for Redistribution. Multimedia and Expo, 2007 IEEE International Conference on , vol., no., pp.48-51, 2-5 (2007).
[187] Tillier, C., Crave, O., Pesquet-Popescu, B., Guillemot, C.: A Comparison of four Video Multiple Description Coding Schemes. EUSIPCO, Poznan (2007).
[188] Lu, M.-T., Lin, C.-K., Yao, J., Chen, H. H.: Multiple description coding with spatial-temporal hybrid interpolation for video streaming in peer-to-peer networks. In Proc. of IEEE Packet Video Workshop, Hangzhou, Zejiang, China (2006).
[189] Liu, J., Rao, S., Li, B., Zhang, H.: Opportunities and Challenges of Peer-to-Peer Internet Video Broadcast. In Proc. of the IEEE, vol. 96, pp. 11-24 (2008).
[190] Herrera, O., Znati, T.: Modeling Churn in P2P Networks. In Proc. of the 40th Annual Simulation Symposium. Annual Simulation Symposium. IEEE Computer Society, Washington, DC, 33-40 (2007).
[191] Singh, J. P., Bambos, N., Srinivasan, B., Clawin, D.: Wireless LAN performance under Varied Stress Conditions in Vehicular Tra_c Scenarios, IEEE International Vehicular Technology Conference, vol.2, pp. 743-747 (2002).
[192] Lee, C. W., Yang, C. S., Su, Y. C.: Adaptive UEP and packet size assignment for scalable video transmission over burst error channels, EURASIP Journal on Applied Signal Processing, vol 2006, pp.1-9 (2006).
[193] Ahlswede, R., Cai, N., Li, S., Yeung, R.: Network information flow, IEEE Transactions on Information Theory, pp. 1204-1215 (2000).
[194] Guo, Z., Wang, B., Xie, P., Zeng, W., Cui, J.H.: E_cient Error Recovery with Netowrk Coding in Under Water Sensor Networks, Journal on Ad Hoc Networks, Vol. 7 , pp. 791-802 (2008).
[195] Mamdani, E.H.: Application of Fuzzy Logic to Approximate Reasoning Using Linguistic Synthesis, IEEE Trans. on Computers, vol. C-26, pp. 1182-1191 (1977).
[196] Gao, M., Ammar, M.H., Zegura, E.W.: V3: a Vehicle-to-Vehicle Live Video Streaming Architecture, IEEE International Conference on Pervasive Computing and Communications, Kauai, HI, pp. 404-424 (2005).
[197] Xie, F., Hua, K. A.,Wang, W., Ho, Y. H.,: Performance study of live video streaming over highway vehicular Ad hoc networks, International IEEE Vehicular Technology Conference, USA, pp. 2121-2125 (2007).
[198] Park, J. S., Lee, U., Oh, S., Gerla, M.: Delay analysis of car-to-car reliable data delivery strategies based on data mulling with network coding, IEICE Transaction on Information and Systems, pp. 2524-2527 (2008).
[199] Chen, H., Sun, B., Zeng, Y., He, X.: An Entropy-Based Fuzzy Controllers QoS Routing Algorithm in MANET, International Conference on Hybrid Intelligent Systems, pp. 235-239 (2009).
[200] Chiang, T. C., Tai, C. F., Hou, T. W.: A knowledge-based Inference Multicast Protocol using Adaptive Fuzzy Petri nets, Journal of Expert Systems with Applications, vol. 36, pp. 8115-8123 (2009).
[201] Wang, Y., Reibman, A., Lin, S.: Multiple Description Coding for Video Delivery, Proceedings of the IEEE, vol. 93, no. 1, pp. 57-70 (2005).
[202] Chou, P. A., Wu, Y., Jain, K.: Practical Network Coding, Annual Alerton Conference on Communication, Control and Computing, pp. 40-49 (2003).
[203] Wang, M., Li, B.: How Practical is Network Coding, IEEE International Workshop on Quality of Service, pp. 274-278 (2006).
[204] Balon, N., Guo, J.: Increasing Broadcast Reliability in Vehicular Ad Hoc Networks, VANET'06, Los Angeles,California, pp. 104-105 (2006).
[205] Evalvid, http://www.tkn.tu-berlin.de/menue/research/evalvid/
[206] Multiple Description Coding (MDC), http://140.116.72.80/~smallko/ns2/MDC.htm
[207] SUMO Mobility Model,http://sumo.sourceforge.net/
[208] SOA4ALL project, http://soa4all.eu/
[209] A.J. Gonzalez, R. Martín de Pozuelo, F. Pinyol, J. Alcober, K.Z. Ghafoor, Context-aware

multimedia service composition using quality assessment, in: Multimedia and Expo (ICME), 2011 IEEE International Conference on, 2011: pp. 1–6
[210] X. Sanchez-Loro, J.L. Ferrer, C. Gomez, J. Casademont, J. Paradells, Can Future Internet be based on constrained networks design principles?, Computer Networks. (2010).
[211] Jun Song, Hung Keng Pung, L. Jacob, A multi-constrained distributed QoS routing algorithm, in: IEEE International Conference on Networks, 2000. (ICON 2000). Proceedings, IEEE, 2000: pp. 165-171.
[212] P.E. Hart, N.J. Nilsson, B. Raphael, A Formal Basis for the Heuristic Determination of Minimum Cost Paths, IEEE Transactions on Systems Science and Cybernetics. 4 (1968) 100-107.
[213] HaiVision, MPEG-4 AVC (H.264) and why, only now, it can save 60% of the network video bandwidth and storage requirements (White Paper), 2011.
[214] Texas Instruments, True System-on-Chip solution for 2.4 GHz IEEE 802.15. 4/ZigBee, http://www.ti.com/lit/ds/symlink/cc2430.pdf, 2011.
[215] R. Ocampo, L. Cheng, Z. Lai, and A. Galis, "ContextWare support for network and service composition and self-adaptation," Mobility Aware Technologies and Applications, p. 8495, 2005.
[216] M. Autili, V. Cortellessa, A. D. Marco, and P. Inverardi, "A conceptual model for adaptable context-aware services," 2006, p. 1533.
[217] L. Yang, J Huai, T. Deng, H. Guo, and Z. Du, "QoSaware service composition in service overlay networks," in Web Services, 2007. ICWS 2007. IEEE International Conference on, 2007, pp. 703–710.
[218] S. Dustdar andW. Schreiner, "A survey on web services composition," International Journal of Web and Grid Services, vol. 1, no. 1, pp. 130, 2005.
[219] S. K. Garg and R. B. Mishra, "TRS: system for recommending semantic web service composition approaches," in Information Technology. ITSim 2008. International Symposium on, 2008, vol. 2, pp. 1–5.
[220] B. Benatallah, Q. Z Sheng, A. H.H Ngu, and M. Dumas, "Declarative composition and peer-to-peer provisioning of dynamic web services," in icde, 2002, p. 0297.
[221] ZhouWang, Ligang Lu, and Alan C. Bovik, Video Quality Assessment Based on Structural Distortion Measurement, 2004.
[222] S. Kandadai, J. Hardin, and C. D Creusere, "Audio quality assessment using the mean structural similarity measure," in Acoustics, Speech and Signal Processing, 2008. ICASSP 2008. IEEE International Conference on, 2008, pp. 221 –224.
[223] T. Thiede, W. C Treurniet, R. Bitto, C. Schmidmer, T. Sporer, J. G Beerends, C. Colomes, M. Keyhl, G. Stoll, K. Brandenburg, et al., "PEAQ-The ITU standard for objective measurement of perceived audio quality," Journal-Audio Engineering Society, vol. 48, no. 1/2, pp. 329, 2000.
[224] ITU-R, "Method for objective measurements of perceived audio quality," Recommendation BS.1387-1, International Telecommunication Union, 2001.
[225] O. Sole Molina, "Multimedia quality assessment," M.S. thesis, Universitat Politecnica de Catalunya, 2009, directed by Alberto J. Gonzalez.
[226] FFMPEG transcoder, http://ffmpeg.org
[227] Dummynet, http://info.iet.unipi.it/ luigi/dummynet/
[228] Burness, L., Eardley, P. y Hancock, R. 2009. The Trilogy Architecture for the Future Internet. *Towards the Future Internet*. (2009), 79.
[229] Guérin, R. y Peris, V. 1999. Quality-of-service in packet networks: basic mechanisms and directions. *Computer networks*. 31, 3 (1999), 169–189.
[230] Hossfeld, T., Hausheer, D., Hecht, F., Lehrieder, F., Oechsner, S., Papafili, I., Racz, P., Soursos, S., Staehle, D., Stamoulis, G.D. y others 2009. An economic traffic management approach to enable the triplewin for users, ISPs, and overlay providers. *Towards the Future Internet*. (2009), 24.
[231] Linder, J. y Cantrell, S. 2000. Changing business models: surveying the landscape. *Business*. (2000).
[232] M. Brunner, H. Abramowicz, N. Niebert y L. M. Correia 2010. 4WARD: A European Perspective towards the Future Internet. *IEICE TRANSACTIONS on Communications*. 93, 3 (2010), 442–445.
[233] Papazoglou, M.P., Traverso, P., Dustdar, S. y Leymann, F. 2007. Service-oriented computing: State of the art and research challenges. *COMPUTER-IEEE COMPUTER SOCIETY-*. 40, 11

(2007), 38.
[234] SmoothIT Project: *http://www.smoothit.org/*. Accessed: 2011-05-19.
[235] Stiller, B. 2009. Telecommunication economics–overview of the field, recommendations, and perspectives. *Computer Science-Research and Development*. 23, 1 (2009), 35–43.
[236] TARIFA Deliverable TWP1.1 D1 Architecture Definition, http://www.i2cat.net/documents/twp1.1.d1_architecture_definition.pdf
[237] Shanbhag, S., Wolf, T. , "Automated composition of data-path functionality in the future internet," *Network, IEEE* , vol.25, no.6, pp.8-14, Nov. 2011.
[238] Seung-Ik Lee; Shin-Gak Kang; , "NGSON: features, state of the art, and realization," *Communications Magazine, IEEE* , vol.50, no.1, pp.54-61, January 2012.
[239] S. Shanbhag, T. Wolf. "Automated composition of data-path functionality in the future internet," IEEE Network, Nov. 201, vol.25, no.6, pp.8-14.
[240] N. McKeown,., et al. "OpenFlow: Enabling Innovation in Campus Networks", In Proceedings of the ACM SIGCOMM Computer Communication Review Vol.38 No.2, pp.69-74, April 2008.
[241] Park K. QoS-sensitive transport of real-time MPEG video using adaptive forward error correction. IEEE International Conference on Multimedia Computing and Systems, pp. 426-432, June 1999.
[242] Nafaa A, Taleb T, Murphy L. Forward error correction strategies for media streaming over wireless networks. *IEEE Communications Magazine* 2008; 46(1), 72–79.
[243] Nafaa A, Mehaoua A. Joint loss pattern characterization and unequal interleaved FEC protection for robust H.264 video distribution over wireless LAN. *ELSEVIER Journal on Computer Networks* 2005; 49,766–786.
[244] Nikaein N, Labiod H, Bonnet C. MA-FEC: A QoS-based adaptive FEC for multicast communication in wireless networks. *IEEE International Conference on Communications*, New Orleans, USA, June 2000, 954–958.
[245] Long L, Ekram H. An analytical model for ARQ cooperative diversity in multi-hop wireless networks. IEEE Transactions on Wireless Communications 2008; 7 (2), 1786-1791.
[246] Chakareski J, Chou PA. Application layer error-correction coding for rate-distortion optimized streaming to wireless clients. *IEEE Transactions on Communications* 2004; 52 (10), 1675-1687.
[247] David Eckhardt A, Steenkiste P. Trace-based evaluation of adaptive error correction for a wireless local area network. *Mobile Networks and Applications* 1999; 4(4), 273-287.
[248] Rouzbeh R, Fleury M, Mohammed G. Adaptive Packet-Level Interleaved FEC for Wireless Priority-Encoded Video Streaming. *EURASIP Journal on Advances in Multimedia* 2009; 2009, 1-14.
[249] Wang Y, Li F. Vehicular Ad Hoc Networks. Springer: London, 2009;1-23
[250] IEEE 802 Committee of the IEEE Computer Society, ANSI/IEEE Std 802.11e, "Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications: Medium Access Control (MAC) Quality of Service (QoS) Enhancements", Nov. 2005.
[251] Kayhan Z. G., Kamalrulnizam Abu Bakar. Inter-Vehicle Communication protocols for multimedia transmission. *IAENG International Conference on Communication Systems and Applications*. Hong Kong, China, March 2010, in press.
[252] Lin CH, Shieh CK, Ke CH, Chilamkurti N.K., Zeadally S. An adaptive cross-layer mapping algorithm for MPEG-4 video transmission over IEEE 802.11e WLAN. Springer Journal on Telecommunication Systems 2009; 42(3-4), 223-234.
[253] Chesterfield J, Schooler E, Ott J. RTCP extensions for single-source multicast sessions with unicast feedback. IETF Draft, version 4, September 2004.
[254] Wiethölter S, Hoene C, Design and Verification of an IEEE 802.11e EDCF Simulation Model in ns-2.26. TKN-03-19 Technical Report Series, November 2003.
[255] NS simulator, http://nsnam.isi.edu/nsnam/index.php/Main_Page, accessed: 25 January 2010.
[256] Evalvid, http://www.tkn.tu-berlin.de/research/evalvid/, accessed: 25 January 2010.
[257] YUV video sequences (CIF), http://www.tkn.tu-berlin.de/ research/evalvid/cif.html, accessed: 25 January 2010.
[258] SUMO Mobility Model, http://sumo.sourceforge.net/, accessed: 25 January 2010
[259] Open Source Fuzzy Logic library (jFuzzyLogic), http://jfuzzylogic.sourceforge.net/html/index.html, accessed 15 July 2012.
[260] Kerner, B. S. (2004). *The physics of traffic: empirical freeway pattern features, engineering applications, and theory*. Berlin, Springer.